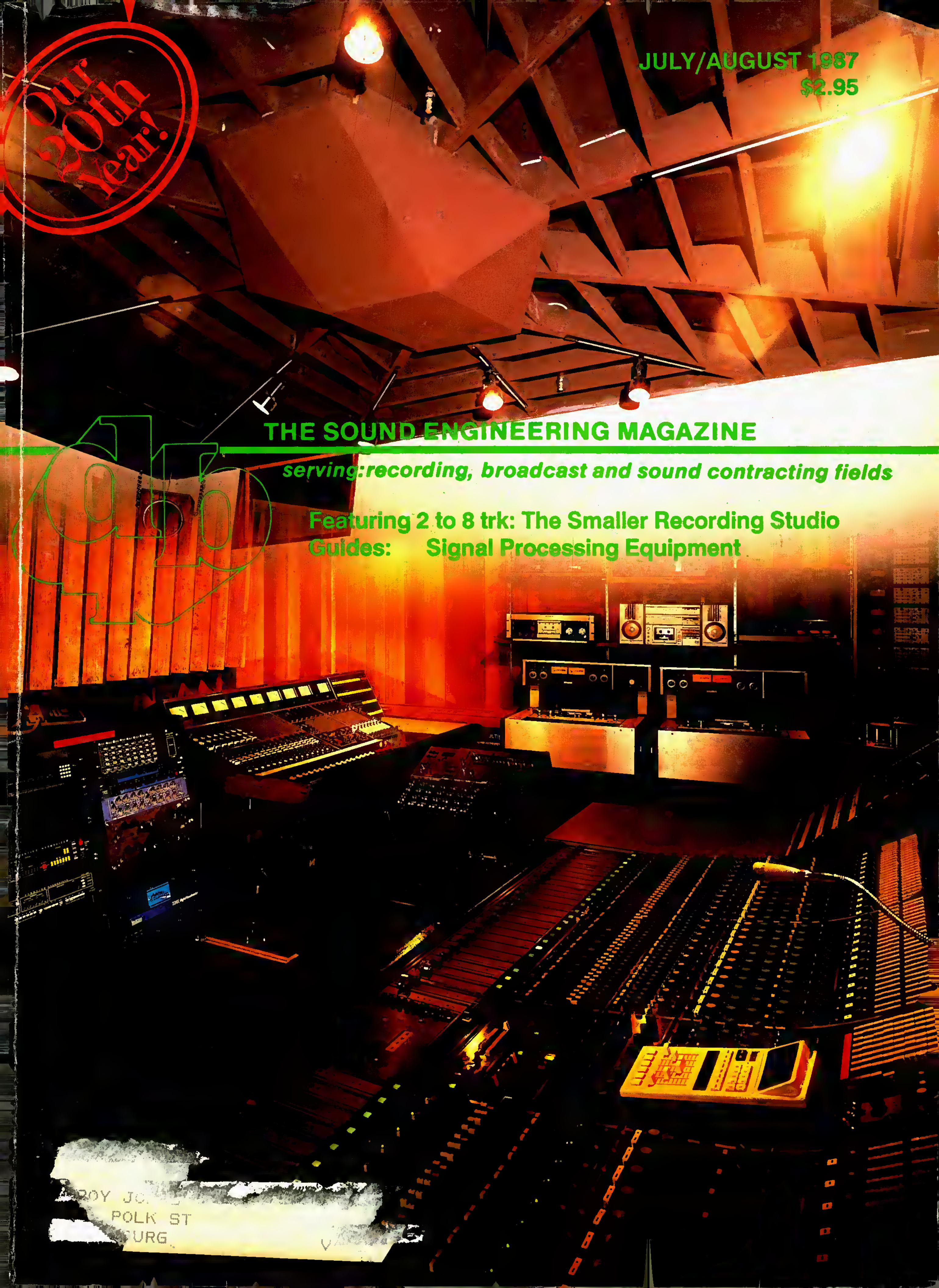

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JULY/AUGUST 1987
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db

THE SOUND ENGINEERING MAGAZINE

serving: recording, broadcast and sound contracting fields

Featuring 2 to 8 trk: The Smaller Recording Studio
Guides: Signal Processing Equipment


Boogie
TAKE CONTROL OF A MAJOR RECORDING STUDIO.
It's time to seize power.
Tascam's Studio 8 puts all the power of a professional recording facility into an 8-track production system that's as easy to handle as your own instrument.
Forget bewildering patch bay jungles. The fully assignable 8 buss mixer sends your signals anywhere you want them with the flick of a switch.
Forget open reel headaches. The unique Load feature loads the tape, senses the end automatically and never lets it run off the reel. And because Studio 8 uses 1/4" tape, you'll never be stuck trying to find someone's rare proprietary tape format.
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But there's something more. Something that's greater than the sum of the features. Startling sound. No other system has Tascam heads. Which means that no other system has the impact and clarity that come from Tascam's thirty years of recording experience.
Quality production is what you do. Do it with a major recording studio. Take control of a Tascam Studio 8.
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About the cover

● When is a picture worth 2,000 words? The composite photo, taken by Jim Cassidy, of the control room and ceiling of Studio A, exemplifies design concepts and how they relate to the quality of product achieved at Cove City Sound Studios of Glen Cove, Long Island.

About the 2-8trk cover

● Page 18. Looks like an ordinary small studio in an ordinary place? Wrong! It *is* a small studio run by Stuart Hollinger and Wes Pacanas, but read more about it and its unusual location in our story **Splash**, beginning on page 19.

JULY\AUGUST 1987 VOLUME 21 NO.4

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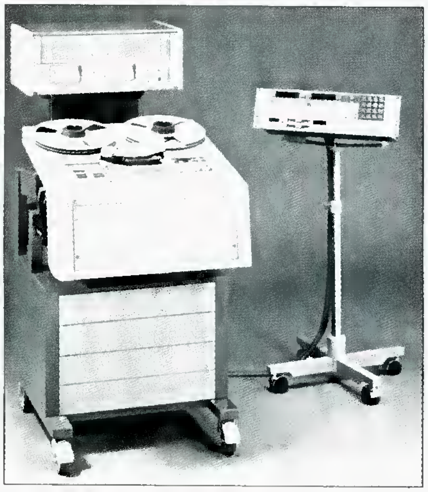
2 to 8 trk the smaller recording studio

Department of Corrections

Subject: **Tascam ATR 60/2T 2-Track Recorder/Reproducer- Lab Report by Len Feldman.**

A correction:

In our last issue, May/June 1987, the Lab Report contained all the correct information and data, but it had the wrong photo of the unit. Here is the correct photo of the Tascam ATR 60/2T.



CALENDAR

• On **Saturday, August 29**, the **UCLA Extension Department of the Arts** will present a special program, "**Professional Practices for Recording Engineers**," for those interested in pursuing recording engineering or who may already be working in the industry and wish to expand their expertise. Among the specific topics to be discussed with industry guest speakers will be an overview of the profession, professional conduct in the studio, trade magazines, trade organizations such as Naras and the AES, legal aspects of recording and job hunting. The program will take place in Room 39 Haines, UCLA, 10am-4pm, for a fee of $25.00.

For details contact:
UCLA Extension
P.O. Box 24901
Los Angeles, CA 90024
(213) 825-9064

• The seventh annual **CMJ Music Marathon** is a four-day music industry conference held at **New York City's Roosevelt Hotel (45th St. and Madison Ave.), October 29-November 1.** Music Marathon consists of panel discussions, seminars, workshops, exhibits, club showcases and the nationally televised 1987 New Music Awards presentation held at the Apollo Theatre on Halloween night.

For information contact:
CMJ Music Marathon
830 Willis Ave.
Albertson, NY 11507

• **SYNERGETIC AUDIO CONCEPTS** announced their fall schedule for their two-day audio engineering seminars. The classes are intended specifically for those seeking a better basic understanding of how to solve everyday audio and acoustic problems. Demonstrations include signal alignment of loudspeakers, measurement of %ALcons and RASTI, the fundamental difference between impulse and energy time curve measurements, and the proven procedure to follow when designing large loudspeaker arrays.

Lansing, MI- August 26-27
Chicago, IL- September 15-16
Kansas City, MO- October 6-7
New York, NY- October 14-15
Denver, CO- September 29-30
Washington, DC- October 27-28
For information contact:
SynergeticAudio Concepts, PO Box 1239, Bedford, IN 47421.

TALKBACK

Dear **db**,

I have some old 2-track masters I dug out of my library. Some are 10+ years old; some are on acetate-based tape.

The trouble is, when I play these old masters, they sometimes make an audible "squeak" on the tape recorder heads. These tapes are quite dry with age. Is there any lubricant I can treat them with to eliminate the dryness causing these audible squeaks against the tape heads?

Thank you.

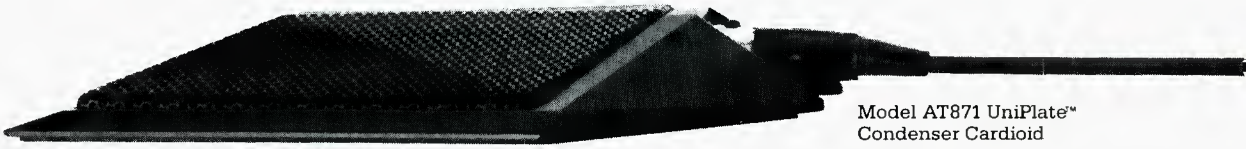
Yours,
Don Spence

Don,
Surprisingly, or not, this problem of "squeak" is not uncommon and, furthermore, is occasionally a problem for many in the post-production areas. However, there are a couple of fairly acceptable solutions. The best one was suggested by Jack Clark, the Revox division shop manager at Studer Revox America. He submitted the following:

Warm temperature and higher humidity of the air can lead to a partial dissolution of the magnetic tape slide coating. This results in increased friction on the tape guide elements and recording heads, which leads to a stronger contamination. Mechanical tape squealing can then occur. This problem can be solved by adding liquid lubrifying fluid.

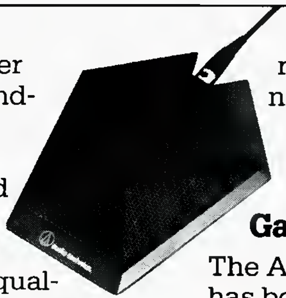
The FilMagic Distributors Group Inc. sell a tape maintenance kit called the FM-LL. In this kit comes a complete array of maintenance fluids (including a lubricant) with a flange-type pylon. Another possible solution comes from the engineers at Tascam Corporation of America. They submitted the following: Tascam manufactures a stainless steel cleaner, which is very high in silicon content, that makes an excellent tape-head lubricant. ■

Who would believe a microphone this flat...



Model AT871 UniPlate™
Condenser Cardioid

could have a curve to match!



If you've tried other hemicardioid boundary microphones, you may have been disappointed in the sound... thin, peaky, and requiring lots of equalization. If so, it's time to listen to ours: the new AT871 UniPlate Condenser Cardioid.

UniPoint Technology at Work

Our experience pioneering UniPoint miniature condensers permitted us to take a new approach to boundary microphone design. We optimized the basic UniPoint cardioid element for boundary use, creating remarkable reach and presence, yet retaining extended high and low-frequency response so vital to natural sound reinforcement.

Outstanding Gain-Before-Feedback

The AT871 UniPlate Cardioid has both the polar pattern and response curve to provide higher gain-before-feedback than you may have thought possible. But better gain-before-feedback and a great sounding element are only a part of the story.

Less Noise Two Ways

By using a very low-mass diaphragm and a case heavier than the others, we sharply reduced sensitivity to mechanical noise. The electronics are audibly quieter as well – a tremendous advantage in typical boundary microphone applications. We also include a low-cut switch to help control acoustic room noise. The AT871 can be powered by an internal battery or from 9-52VDC phantom power.

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The AT871 is solving problems in stage sound reinforcement, church sound, teleconferencing, boardroom applications... even TV and film locations. Wherever great sound is needed...unobtrusively. We urge you to test the AT871 side-by-side with any of the rest. Choose your most critical sound problems. The difference you hear will prove our point.

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MARK E. BATTERSBY

FINANCING FOR NEW INVENTIONS OR BUSINESSES

• Who among us has not dreamed of success based on a new invention or business idea? Unfortunately, profiting from that idea has usually run up against the real world stumbling block of money.

Obviously, a studio engineer or artist with only an idea or invention is not going to be able to successfully apply for a normal business or commercial loan. What, after all, will provide the collateral that most lenders require? Let's look at the decisions that have to be made and take a look at some of the financing alternatives available to the inventor or budding entrepreneur.

The first, and probably most important, decision that must be made is the type of financing that will be sought. The inventor or budding entrepreneur usually has only one asset that can be used as collateral for any borrowing—the idea or invention. Thus, the first decision is whether to give up partial ownership or equity in exchange for the capital or expertise needed to make that idea or invention fly, or whether to take the conventional financing route.

Giving up a portion of the fledgling business or invention is required by virtually every "venture capital" firm. Venture or "risk" capital can best be equated to the formation of a partnership. The inventor or individual with an idea teams up with someone with the money and/or skills needed to get the project off of the ground.

Naturally, just how much "equity" must be given up varies greatly and depends in large part upon the negotiating skills of the entrepreneur or inventor—not to mention the capital or skills that individual can contribute to the new venture.

Since seeking venture capital will mean giving up partial ownership, why not seek the necessary seed or growth funds from private investors? Private investors provide venture or risk capital but often don't have the marketing or business management skills of the venture capitalist. While a private investor will usually demand a piece of the action, the bite will usually be less. Of course, on the negative side, the guidance of an experienced venture capitalist might be more important at this early stage.

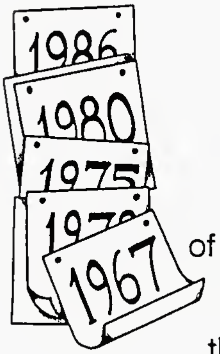
When dealing with private investors, the first rule is to know who you are dealing with. There are all sorts of private investors willing to invest or loan new businesses or inventors money. They range from friends and relatives to professionals who make their living by making high risk loans and investments. Some private investors can be very good partners and some of them are nothing but trouble.

Loans or investments by people you know can be either the easiest or the most difficult to handle. The level of sophistication you'll be dealing with won't be very high and the final terms of the loan or investment may not be very severe. It is important for both peace of mind and the survival of your venture that you accept loans and investments from friends, relatives, associates—and your employer—on the most business-like basis possible. This means a formal agreement or contract.

When seeking private sources of financing, an employer, possible supplier or even the end user of your invention or service might provide seed money for a start-up. If direct financing or investment is not a possibility, why not tie your venture to their existing operation. They put up the capital and the marketing and management know-how, you the invention, idea or time involved.

If, however, you are mining friends, relatives and associates for start-up capital, it may be easier to locate a co-signer than a lender or investor. That's right, in exchange for partial ownership or other monetary considerations, an individual—or busi-

DB INDEX IS HERE!



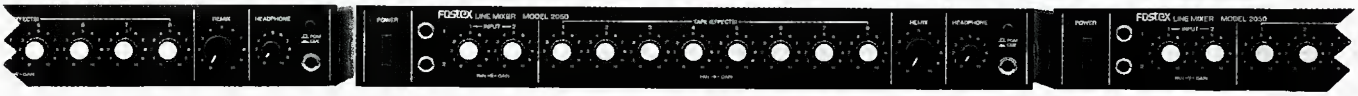
See page 56 in this issue for a coupon for the cumulative index of db Magazine, available from the years 1967 to 1975, and from 1976 to 1985.

This is great.

Model **450**, 8 x 4 x 2 Mixer, features phantom powering for professional condenser mics, in-line monitoring, solo, a dedicated stereo mix buss, stereo effects send and four switchable LED bar graph meters. More routing flexibility and performance gusto than anything in its price category. **$1095.00***

This is even greater.

Model **450-16**, 16 x 4 x 2, is a 450 with eight more inputs. The perfect 16-track production console with sonic quality you'd expect to cost three or four times more. Nothing else even comes close. **$1995.00***



And these are the sonic equivalent of duct tape.

Model **2050**, 8 x 2 Line Mixer, will save your day when you suddenly need: another cue mix; a keyboard sub-mix or drums sub-mix; a separate monitor feed; a quick vocal reference (there's a mic preamp input, front panel). For just about any conceivable signal routing problem, latch onto a Fostex 2050 or two, and the problem is solved. These little sonic lifesavers are **$260.00*** Each. No audio tool kit should be without one.

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Great mixers at great prices, the best of both.

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*Suggested retail prices are subject to change without notice.

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ness—might be found that will act as a co-signer for a conventional loan. In essence, you will be using someone else's credit to borrow the money necessary to promote your invention or to develop your idea.

An entrepreneur or inventor often borrows on personal assets. The classic example is the equity that has built up in your home. A second mortgage can help convert that equity into ready cash although at a higher interest rate than was paid for the first mortgage. The second mortgage route can be especially attractive in these days of rapidly escalating real estate values.

Borrowing against life insurance policies—or using them as collateral for a loan is not as attractive as it once was. However, it is still an effective strategy and guarantees that no outsider will be able to leverage his equity interest to the point of excluding you, a real danger every time an outside investor is brought in and given too much control of the venture or invention.

As already mentioned, most bankers will flatly refuse to lend to a new business without collateral or a track record. Those rejections, however, provide the basis for approaching the Federal government for needed funds. There is the Small Business Administration to consider, for example.

The Small Business Administration (SBA) is an agency of the Department of Commerce whose sole purpose is to help "small businesses grow and prosper." The SBA provides consulting services, technical assistance and advice. The SBA will also assist a small business person to secure government contracts. Its most popular program, however, is money.

SBA financial assistance is normally provided in the form of a loan-guarantee in which the SBA insures a loan through a commercial bank—up to $350,000 or 90 percent of the loan, whichever is less. The SBA may also "participate" in a loan along with the bank—up to $150,000. In fact, if bank financing is not available, the SBA can even loan money directly to the business in any amount up to $100,000—but funds are extremely limited.

On a local level, the SBA licenses privately-owned Small Business Investment Companies (SBICs). These supply both long-term finance and venture capital to small ventures for expansion, modernization and "sound financing" of their operations. Management assistance may also be provided whether wanted or not.

In many areas, another source of government or quasi-government financing is provided by developmental corporations or agencies. These development organizations offer such incentives as tax breaks, loans, surveys, employee training, site or facility location, advice and management assistance to businesses that provide employment in their area. The development corporation may be sponsored by a local utility company, a town, a region or even a state. But they all have one purpose and that is to create more jobs for their areas.

Anyone attempting to profit from a new invention or start-up a new business venture should do his or her homework carefully. There are many sources for that all-important capital or financing. All can provide capital and all have certain drawbacks such as high interest cost, paperwork burdens or loss of ownership. But they are all in the business of lending or investing money and thus are all in competition. Shopping is definitely in order.

The key to asking for or raising money in the right way is information. If the lender or investor doesn't have the information they require (or think that they need) to make a decision, there is little hope of getting the needed money. Recognizing what information will be needed and supplying it without being asked is also the sign of a good manager, something every lender and investor is looking for.

When a company decides to sell its stock nationally, it is required to file a prospectus showing the details of its financial position and other important data. Copies of this prospectus must be provided to all investors to let them see where the company stands.

This isn't a bad idea for a new business or invention. What is needed is a strong statement of where you are, who you are and where you are going this business plan is a fairly large document.

The business plan will include more than financial statements. It should also include a survey of your potential market, a statement of just who you (and other key participants) are and how you plan to grow and prosper. It will also include budgets and projections for coming years.

The financing route an investor or entrepreneur follows depends on a great number of factors. Will giving up equity pose a problem? Is marketing advice or management skills required from an investor or lender? Can ordinary financing be obtained with or without a Small Business Administration guarantee? What combination of financing makes the most sense for your venture or project?

Most importantly, the bottom line may be not so much what the inventor or entrepreneur desires, but rather what financing sources are available for this project—and at what cost? Financing is available, so begin to prepare that all-important business plan today. ■

Ad Ventures

BRIAN BATTLES

• It is often quite advisable to add distortion to your recordings. We all know that the word *distortion* carries an evil connotation in the field of recording and sound engineering, but it is rare for a producer or engineer to avoid purposely introducing distortion to almost any recording. You see, strictly speaking, distortion is any deviation from the original characteristics of an audio signal as it passes through electronic and/or acoustic media. Anytime you make adjustments involving equalization, reverberation, compression, or other forms of processing, you are actually distorting the original sound. (Technically, I suppose volume control and amplification even could be considered forms of distortion, since they involve artificially modifying the amplitude of a waveform, but let's not go to extremes.)

In the last *Ad Ventures* we touched on some of the types of equipment you might wish to include in your studio configuration, and beginning with this issue we are going to devote a few installments to investigating how signal processing devices can be used in the production of radio commercials.

It is important to keep in mind just what is the purpose of producing radio commercials. Obviously you want them to help sell something, but if you analyze things closely you will understand that this is the indirect result. The immediate and primary goal of a broadcast advertisement is to be heard. If no one hears a spot, it may as well not be recorded, and that certainly won't help you earn a living. So, if your work is destined for transmission through the ether, you ought to know a bit about the technology involved.

First off, understand that in many ways *radio = distortion.* There is a widely accepted notion in the broadcasting business that the greater the apparent or perceived loudness of a station, the more people will listen. Apparently studies have indicated that the average schnook somehow believes that the loudest station on the dial is the one that comes in the strongest, plays the best music, offers the most accurate news, features the finest air personalities, and provides the source of eternal youth, happiness and sexual prowess. Of course, most listeners have no idea how much clipping and compression must take place inside the arcane circuitry of a transmitter in order to provide this added "volume," and radio station owners certainly couldn't care less about quality as long as the ratings are up.

What equipment will your masterpieces be played on? I think you had better grab your crying towel when you read this. Virtually all radio stations take commercials produced on open reel dub and dub them onto broadcast cartridges (carts, to those of us in the biz) which closely resemble the old primeval 8-track tapes are slightly more serious doodads, however. First of all, they roll at 7 1/2 in./sec. (as opposed to the 3 3/4 in./sec. speed of 8-tracks). Secondly, they don't carry their own self-contained pinch roller; they have a little hole in the bottom of the shell and the roller in the record/playback deck (cart-machine) pops up when the tape is running. Thirdly, they are supposedly loaded with halfway good tape and are designed to be rather rugged. A typical cart may be played over a hundred times a week for years before it gets so worn out that somebody finally tosses it in the barrel. Most cart machines are built for nothing more than sheer durability, and few are equipped with any sort of noise-reduction circuitry. The audio specs of broadcast cartridge technology can be politely termed "reasonable" at best, and that's normally a valid description when they're still fairly new. I understand that better equipment and digital technology are beginning to make inroads in this area, and the manufacturers are sometimes capable of constructing units with respectable numbers, but in real life most disk jockeys and production people abuse the machines and the carts to the extent that their fidelity falls short of high. Based on my experience, few of the people who operate the equipment in the radio industry even know the general definitions of terms like *headroom, saturation, print-through, bias*, and so on. I know, I know. Now sop up those tears and let's move on.

What about radio's frequency response? How does 50Hz-15KHz ±3dB grab you? The joke's on you, because that's the *potential* technical limit of radio (not just FM, but AM, too, believe it or not). We live, however, in a slightly less than perfect world, and I will publicly apologize in this column to any station whose chief engineer can prove to me beyond reasonable doubt that his or her station cranks out those specs all the way from the cart machine to the receiver on a normal, routine daily basis (no fair twiddling and tweaking during those late-night "proof of performance" sideshows!) So, forget the thundering bass and shimmering treble, 'cause they ain't there. As to hiss and noise, I have yet to see a properly Dolbyized or dbxed radio station. Believe me, cart machines running at more or less 7 1/2 in./sec. with weekly (!) head cleanings and monthly demag sessions are not exactly down in the sub-basement regarding noise floor. *Most of the stations I have worked for did not even do that much maintenance!*

(My humble apologies and kudos to the few broadcasters that follow decent rules of cleaning and maintenance. I know what a battle it is to get disk jockeys and production prima donnas to pick up a cotton swab and a bottle of non-consumable alcohol. I gave up and did it myself when I was a production director. Heck, the live version of *Free Bird* is good for three cart machines and a reel-to-reel or two—including pinch rollers. Psst...wanna buy some black Q-Tips?)

We can see, then, that an extraordinarily clean audio source far exceeds the technical capabilities of the broadcast facility, so don't wring your hands because you can't get them on the latest high-tech, obsolete-tomorow CD-Digital-Neutron-Laser-MIDI-Photon Torpedo tape deck. I even wonder why stations bother to make such a fuss over their playing of sacred compact discs, since they just go ahead and compress the dynamic range to death. And if their argument is that CDs don't skip or get scratchy, all I can say is, "Bull droppings!" I have lost count of the number of times I have heard compact discs go berserk on the air, with their unique electronic conniptions spewing out noises never before heard in the Australopithecine days of normal phonograph records.

So we know that we want our commercials to sound loud and that we don't need to worry about dynamic range, inaudibly superior frequency response or a little noise. Just good old standard professional production techniques will do fine. That means clean the heads, don't go too much into the red and never record over a splice.

Two good methods of cranking up the loudness without wrapping the VU needles around the pin are: 1) manual gain-riding and 2) electronic compression. Both can be used concurrently to achieve fine results. Now, nobody can possibly twist their wrist at anywhere near the speed necessary to hammer a tape to a hair below saturation without blowing off at least a few dozen peaks per second, so a good compressor/limiter is really a necessity. Most of the reputable manufacturers have models that work quite nicely. For example, Orban, Ashly, Valley, Symetrix, dbx and many others make terrific units that will not require you to apply for a second mortgage. Do yourself a favor, though, and go easy. Read the operation manuals and be careful with the attack and release knobs. Go ahead and push the ratio, though. Anything less than 10:1 or 15:1 barely meets the stations' own squashing efforts, so go for it. I run at least 25:1, or even ∞:1 for total flattening, as long as the release time is well over a full second. My only exception is on straight, dry voice-over spots or ads with touchy sound effects. Practice and experiment. If you also choose to ride gain manually, watch out for excessive *ducking*. This is that obnoxious "pumpy" sound created as the music bed leaps up to fill every gap between the announcer's syllables. That was okay in the screamin' Top 40 days of the 1960s, but in 1987 it can be really irritating. Smooth but tight compression is the rule to follow in radio commercials and blue jeans.

While we're on the subject of compression, we ought to take a moment to discuss stereo production. What? You mean you thought...that is, you naturally assumed that...in other words, it's okay to cut most radio spots in *mono*?! Absolutely. As a matter of fact, most stations don't even own stereo playback decks for their commercials. Which is just as well, in many cases. You see, stereo production can be your own worst enemy when it comes to broadcasting. Here's why: the compressor/limiters at a radio station are often strapped between channels, meaning that what triggers the left side also affects the right side. I found this out on an early project I finished at a New York radio station. This was the first time I had been at a place that ran stereo carts, and me being Mr. Creative decided to whip up a slick little two-voice job with me carrying on a conversation with myself across the tracks. When the big moment came and it hit the airwaves, I squirmed through a sixty-second game of audio ping-pong where it sounded like a pink noise generator was being carried from one speaker to the other by Speedy Gonzales. Not to mention the fact that my dulcet-toned voices were around 3dB down on each side. But that's not all—when I gave it a second listen on a mono receiver, I thought for sure that I had somehow tuned into an AM skip out of Seattle. Imagine a clear FM jock in afternoon drive-time coming out of a Steely Dan set and segueing into a sound like late night shortwave from Madrid. I learned the hard way that without gently nudging a correctly balanced pan pot, that nutty old L+R/L-R multiplexing scheme wreaks havoc with pots and pans, a proper pan pot is designed so that at the 12 o'clock setting it doesn't merely sum the levels from both channels like the balance control on a hi-fi set, but simply attenuates the signals evenly as it is rotated in either direction.) So now I say, "Watch it!" That far out lead guitar solo or ringing phone sound effect panned way over to the right or left is going to sound like a real far *away* lead guitar or a telephone booth out in the parking lot by the time it reaches the listener's ears. When you hear that (and by then it's too late), you will wish you had foregone the artistic stereo folderol and simply gone for the jugular with some hardball distortion techniques like compression.

Do not consider the term "distortion" to be a necessarily dirty word. Try a little here and there; it gives marvelous results when used in certain broadcast situations, and it can help you compete with the big boys. Next issue we will probe around the innards of some other sonic distortion techniques that can spice things up quite nicely.

TALKBACK MIC

I would like to acknowledge some dedicated professionals: Wayne Gerbrandt of FastTrack Recording Studios in Denver, CO; Bill Woodrome of Sound System Supply in Woodville, TX; and Bert Yanowich of Balajag Productions in Grand Junction, CO. Keep those letters and phone calls coming—I love one-to-one tech talk, it's great for my ego, and it sure can't hurt when it comes time to ask the publishers of *db* for a raise...Big thanks to Norm Simmer and Glenn Watts at ListenUp in Denver as my current number one pro sound sales/support dealer...I would like to express my gratitude to David Schwartz of Compusonics in Palo Alto, CA for answering my dumb questions and to Regis Dahl of Golden, CO for graciously letting me and my henchpersons invade his gorgeous home to check out an exciting new piece of computerized audio technology...Also, get a load of what's happening with Tom Russo and Paul Lombardo at L & R Productions in East Hartford, CT. They're doing some of the hottest radio and TV commercial production in the Northeast...Hello to former Rhinestone Kal David, somewhere in southern California...Speaking of the Golden State, thank you Greg Diaz of Tascam for some tech help...And a big hello and tip of the cans to Dan and Lisa Clawson of ClawsOn Productions in Boulder, CO. Listen for Dan's sax on the new digital Pure Prairie League album, and watch for his amazing top secret electronic wind instrument to turn reedmen on their ear soon...Oh, yeah: why isn't superproducer Jim Mason buried with album projects right now? I guess nobody's looking for a hit record at the moment. ■

KENT KLAVENS

PRODUCING UNSIGNED ARTISTS: NEGOTIATING FOR MAXIMUM SUCCESS

• Suppose you're an engineer with talent, and you're tired of working for producers possessing only equal or lesser skills than you, but who get all of the recognition (and most of the money). Or suppose you're a producer "for hire," working solely for artists signed to record deals and receiving only a few royalty points and minimal cash advances. What can you do to improve your status and financial success?

In the last issue, I discussed producers' agreements from the perspective of artists and songwriters. This article will cover the type of deal potentially most advantageous for producers, as well as for engineers who wish to become established as producers. I'm referring to the "production deal."

Under a production deal, the producer operates as a production company, signing and producing unsigned recording artists and "shopping" or promoting the artist and the finished master recordings to record labels. In this situation, the artist is employed by the Producer. An interested record company will then sign the act "through" the production company with an "all-in" deal, as the production company then "furnishes" the services of both the artist and a producer to the record company (and often also sells or licenses the master recordings produced before the record deal).

Don't let this idea intimidate you. The "production company" is mostly a concept. Although certainly there are some laws you will have to follow in operating any business, don't mistakenly visualize some stuffy office building with a crew of secretaries and bookkeepers on the payroll. Most production companies operate out of a recording studio and might involve only the producer and an office assistant.

Under what circumstances would an artist accept a production company situation? Many artists have neither the funds nor the expertise to either produce or pay someone to produce master recordings suitable to attract the interest of a record company. Your talent, as well as your access to suitable recording studio facilities at little or no cost to you, makes you an extremely valuable commodity to them. For these kinds of artists, this places you in a position of substantial negotiating power.

How does your financial participation in a production company situation differ from working for artists in exchange for advances and royalties? First, of course, if the record company wants to use the master recordings you've produced at minimal cost, there can be a profit when they purchase these masters from you. In a production company situation, *you* own the masters, not the artist. Most importantly, however, the production company receives all of the advances, royalties and other payments from the record company for both the producer and artist, paying the artist only whatever percentage has been agreed upon under the production deal. This should always provide a larger share for the producer than would be received in a situation where the artist has hired the producer.

Normally, if an artist has hired *you*, they will offer you "points" of their royalty in exchange for your services, either with or without additional cash advances. Under these circumstances, if the artist gets a record deal:

1) you will receive two to four "points" of any royalty paid to the artist which is based on some version of the retail price of recorded products sold (approximately double that number for any royalty based on some version of the wholesale price); and

2) you will most likely be paid only for the use of the master recordings you produced, so your share of any album will be reduced substantially if other master recordings are contained on that album or if songs you produced are substantially re-recorded, and you will receive nothing at all for later albums.

Under a production deal, you are taking the risks by charging no production fees up front and by providing recording studio time the artist could never afford. You are also finding a record deal. Therefore, you deserve a much larger reward for your efforts. In this situation: 1) because the artist is working for you, you will participate in royalties from *all* recordings made by the artist during the life of the agreement with the record company, *whether or not you personally produced the master recordings involved;* and 2) you can usually negotiate that you will retain from 50 percent-60 percent of the total producer and artist record royalties paid by the record company (of course, if you do not produce particular tracks, you will ordinarily be responsible for hiring and paying some other producer out of your portion of the royalties).

In a deal with a major label that pays royalties based upon some version of the retail price of recorded products sold, the "all-in" royalty (artist and producer combined) will be in the range of 11 percent-13 percent for the first album produced under the deal (depending on the computations in the recording agreement, for albums receiving full royalty rates, this amounts to $0.75 to $0.90 per album, perhaps rising to the 14 percent-16 percent range, or $1.00 to $1.25 per album in later years of the agreement). Therefore, even in a 50-50 artist/producer split under the production company deal, you will retain 5-1/2 to 6-1/2 points, a royalty rate you would never have approached otherwise until you reached superstar producer status. If you decide to hire another producer for three or four points, you will still be able to retain 2-1/2 to 3-1/2 points just for arranging the deal!

Another major financial advantage to you will be the split of excess production monies. Most major record labels provide a recording "fund" for each album. For newly signed, mainstream pop, pop-rock and r&b-pop acts, this sum can be in the range of $125,000 to $150,000, (and perhaps greater if more than one major label has actively pursued your artist during negotiations). Normally, if you are merely a producer hired to produce the act, you will receive only a specified sum as your advance (recoupable from your royalties). In that situation, it doesn't matter how much money you save in production costs, because your advance has already been determined.

Under a production company agreement, you will be able to keep from 50 percent-100 percent of all excess production monies (most labels will pay any leftover portion of the recording fund upon delivery of the particular album). So if you produce a $75,000 album with a $150,000 recording fund, you will probably keep a substantial portion of the remaining $75,000 (assuming you were able to deliver a "commercially and technically satisfactory" album to the record label for sums you actually spent). The benefit to you is potentially even greater in later years of the agreement with the record company, when the recording funds for albums will likely be in the $200,000 to $500,000 range.

When any of your prospective recording artists also primarily write their own songs, another potential financial incentive for you in production company situations might be to obtain some portion of the "music publishing" rights to all or some of these mutual compositions. These rights involve ownership of some or all of the "copyrights" to these songs, as well as the power to issue "licenses" (permissions) for all uses and to collect money from such uses. Although some production companies are able to obtain all music publishing rights from their artists, many songwriting artists will relinquish no more than fifty percent of the publishing rights (a "co-publishing" agreement), and will only grant such rights as to songs recorded and released under the agreement between the production company and the record company.

The basic mathematics for music publishing are fairly simple. From each dollar of gross income, the songwriter is normally paid 50 percent as an absolute minimum. This is the "songwriters'" share, which is paid regardless of who owns the music publishing rights or how such rights are split. The remaining 50 percent is the "publishers'" share, which is either wholly retained by the music publisher or split in some manner under a "co-publishing" arrangement like the one described above.

Without going into a detailed explanation of copyright law, its legislative history or the bargaining power of record companies, suffice it to say that most major record companies are now paying music publishers three and three-quarter cents per song per copy of a record sold, with a maximum of thirty-seven and one-half cents per album sold, for the rights to "mechanically reproduce" musical compositions on records. If an album contains ten musical compositions written solely by artists who have assigned all of their music publishing rights to you, and if that album sells one million copies, your music publishing company will receive an additional $375,000 in mechanical royalties to be split equally with the artist.

Aside from mechanical royalties, as to any hit singles that receive substantial radio station airplay, the publisher will receive large payments from a "performing rights society" (usually ASCAP or BMI) for these "performances." A major, #1 chart single could generate additional payments of $50,000 to $75,000 or more to *both* the songwriter and music publisher (ASCAP and BMI pay songwriters and publishers separate, equal shares), depending on how long the song was at or near the top of the charts.

There are many other areas of music publishing income that are beyond this discussion. You should be aware, however, that this will be a highly lucrative source of income if your artist's songs achieve major success in the music industry.

Aside from the potential financial rewards outlined in this article, it should be obvious that the greater the success of an artist signed to your production company, the greater your status and power with respect to record labels and other artists. If you pick the right artists and negotiate intelligently to maximize your position, you may someday realize dreams you never thought possible. ■

Kent Klavens is an entertainment industry lawyer with offices in Los Angeles.

Book Review

JOHN EARGLE

A Review of Bruce Bartlett's Book, "Introduction To Professional Recording Techniques"

Tutorial texts in sound recording have been available for about the last ten years, and by present count, there are no more than half a dozen that are routinely used for teaching. Bruce Bartlett has now added another one to that limited list, and a welcome one it is, too.

The book is part of the John Woram Audio Series published by Howard Sams & Company, and it reflects, as all the available texts do, the particular tastes and experiences of its author.

The book is large format paperback and very well laid out. There are copious drawings and photos, and Bartlett covers his chosen subjects thoroughly and lucidly. The book generally stresses practice and technique as opposed to theory; this is appropriate, inasmuch as there are plenty of other books with the opposite orientation. As such, the book is very easy for the recording novice to read and comprehend.

The book begins with a broad description of the recording and reproducing chain, explaining in clear terms what the basic steps are. Of special interest here is a listing of the many levels of transduction and signal modification which can exist between input and output.

Next comes a short description of what it takes to get a home studio up and running, including the costs involved. This is followed, appropriately enough, by a discussion of rudimentary studio acoustics. Again, the discussion is based on the economies of home studios.

Next comes monitoring, speaker placement and power requirements. Headphone monitoring and cueing systems are also discussed in this section.

Then comes a very useful section on hum prevention, shielding and RFI prevention. These are subjects that are not often described in recording books, but which are obviously of importance. The coverage stresses diagnostic methods and reliable fixes.

Microphones are referred to by transducer and pattern type. Basic electrical characteristics, such as sensitivity, impedance, noise and distortion are examined, along with a discussion of useful microphone accessories. Basic microphone selection for particular recording applications is covered rather specifically, but without prejudice.

The author stresses the number of microphones to be used for which kind of activity, and the principles of stereo micing are succinctly covered. Problem areas such as noise, leakage, off-axis coloration and the like, are also included. In many ways, the basic treatment of microphones and their usage forms the heart of any book on recording, and Bartlett has done it well indeed.

Then comes a long chapter dealing with specific mic techniques for specific instruments or groups of instruments. For the most part, this section is based on common practice, and the explanations are lucid.

Tape recording is discussed from the practical points of view of general operating principles, maintenance, alignment and operating levels. Noise reduction is covered, as is general information on editing and tape storage. The digital recorder is not much more than hinted at, and this is appropriate, considering that it will be many years before it is a significant factor in home recording.

Signal processing covers the principals of usage for the whole range of equalizers/filters, compressors/limiters/noise gates, and the various time domain manipulations of delay and artificial reverberation. This leads right into a discussion of consoles, rudimentary console architecture and short descriptions of basic functions.

With all of the basic hardware out of the way, the author continues with details of session procedures, including setup, breakdown, session protocol, overdubbing, program assembly and special effects. The spoken word is given its own chapter, as is the new technology of sampling, sequencing and MIDI interface. On-location recording of both popular and classical music is discussed in terms of basic setup, operational aspects, manning requirements and musical aspects.

Final sections of the book deal with judgements of sound quality, how we listen. Short appendices deal with decibel notation, SMPTE time code, and listings of schools, references and other literature.

If the book has a particular aim, it is toward the neophyte recordist who may or may not have access to formal training or apprenticeship, and who may have to educate himself as he grows in his art. For this purpose, the book is quite thorough, imminently readable and hits its mark squarely. ■

Editorial

In My Opinion

The following guest editorial is from Drew Daniels. You will, of course, recognize him as a frequent contributor to these pages. He is also with JBL as Applications Engineer, and is thus well-qualified for these solicited opinions.

Congress, well-meaning as it might be, is debating and voting on technological issues with insufficient technical information, or with information biased by special interests. This sort of thing has become the symptom of America's headlong tumble into the status of second-rate technological power. Our Congress has been asked to consider a bill that would require tape machine manufacturers to put a device into new tape recorders to prevent pirate recordings and home taping of the ultra-high quality digital disks (Senate bill S.506) and a bill to outlaw new recording technology altogether (House bill H.1384). The house bill is a total absurdity and does not even deserve discussion, but the Senate bill is even more dangerous to the future of recorded music.

On the surface, preventing pirate taping seems like a reasonable effort, except for the fact that victims of the last forty years of pirate and home taping will not benefit from it.

Audio scientists and engineers have never consciously tried to make the sound quality of their products worse. Now, record company and Compact Disc manufacturing interests are seeking to undo the improvements achieved in audio and return us to the sound quality levels of the 1950's and to a second-rate position in the world of audio technology. What's ironic about this disaster is that it is easily avoidable.

The details are somewhat technical, but it boils down to this: the record companies want tape machine manufacturers to use a microchip that would recognize the absence of a band of music that had been removed from the original recording, a band roughly equivalent to removing blue-green from your television. This is not necessary. By design, it is alternately possible to encode the minute man-made background noise, called dither in the digital recording process, and use a microchip that would respond to this code. This latter approach would not alter sound quality in any way.

The question is clearly, should we allow reactionary special interests to stifle the inevitable advance of technology or ruin what we have achieved with great difficulty because they are in a hurry to protect themselves against losses they have already been subject to since the introduction of the tape recorder more than forty years ago, or should we pursue our earnest efforts to create viable recording and playback means that continue to provide the public with the best the world and our Nation's audio professionals can provide.

As a record producer, I am naturally concerned that dishonest people might steal my product by illegal taping, but I am also troubled to think that purely financial special interests want to alter my artistic intentions and allow the public only products inferior to what I actually produce. Please help me. Vote against the proposed measures to require the inclusion of this travesty. Vote NO on Senate bill S.506 and send the message that America wants to go forward, not backward with technology.

We at **db Magazine** completely concur with Drew Daniels' letter, save one critical point. We do not believe that anyone needs any form of copy protection, since, the long-claimed losses suffered by elements of the record industry have never been substantiated. The same cries for protection were made when the analog cassette was first introduced, but in fact, the cassette has proven to to be financial savior to record sales.

The excuse that the new technology of DAT is so good, that copies will be undetectable is only a new excuse for an old demand.

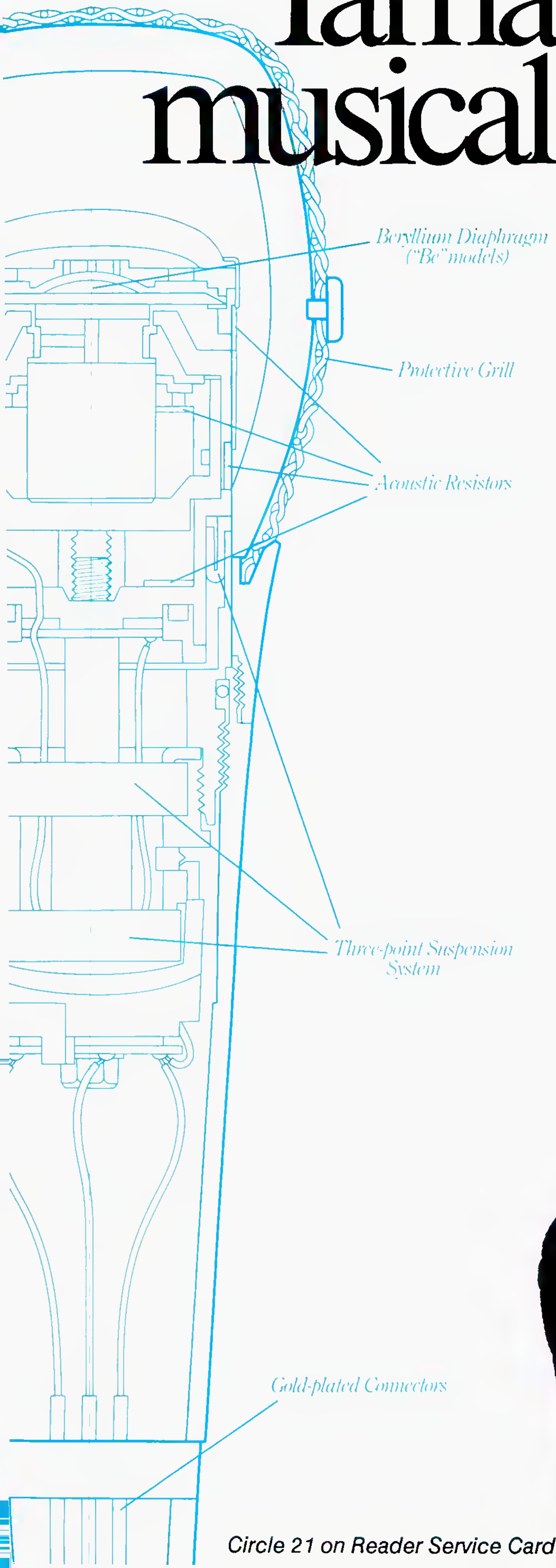
We can envision a $300 DAT player down the line. But we can't envision an inexpensive digital tape that kids will buy. That tape will continue to sell in the ten dollar range. Why would anybody use expensive DAT technology when they already have inexpensive analog?

The fact is that this federal bill will soon lead to others so that ultimately, all consumer audio tape decks, audio and video, would ultimately be required to have the cutout chip in them. That is what the RIAA and its member proponents really want anyway.

Will the musician get more royalties from these proposals? We don't see how, but we do see possibly more profits for the record companies, since as usual the consumer will have to pay more for the "protection" technology-added products of the future.

LZ

Yamaha's newest musical instruments.



For years, Yamaha has been making musical instruments that allow performers to express what they feel. Our new line of MZ Series professional dynamic microphones continues this tradition.

The three-layer laminated beryllium film used in the diaphragms of the MZ102Be, MZ103Be and MZ105Be offers greater control over piston movement for heightened presence and accuracy throughout the entire frequency range.

Throughout the entire line, a superior acoustic damping system makes use of a mesh screen and photo-silkscreen technology. The benefits are extraordinary long-term stability and absolute uniformity. This uniformity can be especially important when matching pairs of mics for stereo.

A tight cardioid pattern provides excellent off-axis rejection for superior feedback suppression.

To reduce handling noise, all MZ mics have a unique three-point floating suspension system. And a special windscreen with three times the impact resistance of conventional types. So you know it can take a pounding.

We even use gold-plated audio connectors.

But when you listen to Yamaha MZ mics, you hear more than the result of advanced technology. You hear a one-hundred-year tradition of making music.

For complete information, write Yamaha International Corporation, Professional Audio Division, P. O. Box 6600, Buena Park, CA 90622.

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STUART HOLLINGER AND WES PACANAS

Splash

A studio in paradise? Almost, when you learn the where and what of this MIDI-equipped operation.

Splash Recording Studio is located on the idyllic island of Kauai, Hawaii. We are local resident Hawaiians and long time performing musicians, as well as owners and operators of the studio. Our concept of Splash was to create a good quality affordable studio. By affordable, we mean not only for professional musicians, but any and all types of audio recording. Splash is Kauai's first MIDI equipped studio.

Splash is a family run business with wives Mika and Patti attending to reception, bookkeeping and all those time-consuming front office details.

Wes is master engineer and computer expert. Through his ingenuity the control room is equipped with some amazing MIDI programmings! Just about anything you could desire, he can create! Stuart is assistant engineer and in charge of management and public relations. Our combined efforts create a relaxed, proficient, smooth-running atmosphere.

Being set up as a MIDI 8-track, we can reach a wide and varied range of clientele. Splash offers discounts to senior citizens and aspiring young students. We feel the studio should be accessible to everyone, not just a select few. Splash is also involved in advertising, radio and television commercials and video soundtracks. We provide our customers with personalized music and customized jingles. Financially, a studio only limits itself by not making full use of all its facilities.

BUILDING SPLASH

Due to patient planning and forethought, we have run into very few problems. The studio was designed and built around an already existing structure. The design is by us, and the actual construction was supervised by Wayne 'Yama' Yamaguchi. Having had a direct hand in the construction resulted in tremendous savings. Yama's expert craftsmanship can also be seen in the fine wood finishing throughout the interior of the studio. Splash was designed to be practical, functional and pleasing to the eyes.

Figure 1. The studio floor plan.

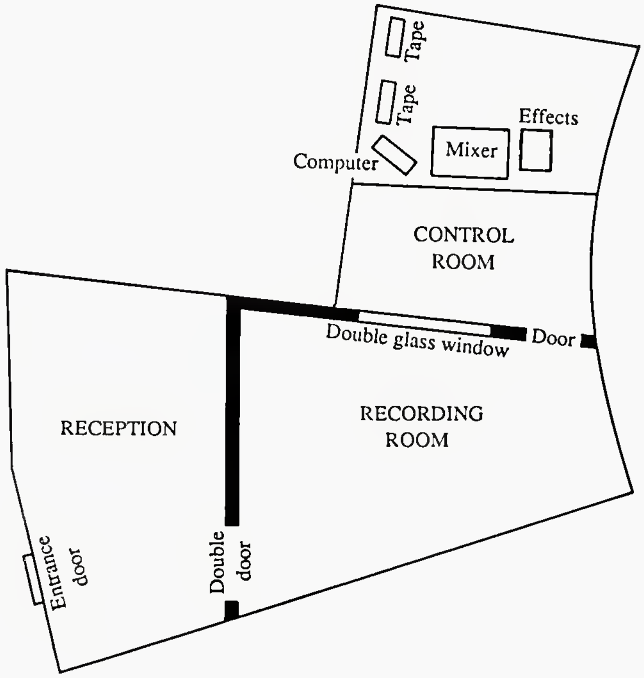


Figure 2. The keypad on the custom remote unit.

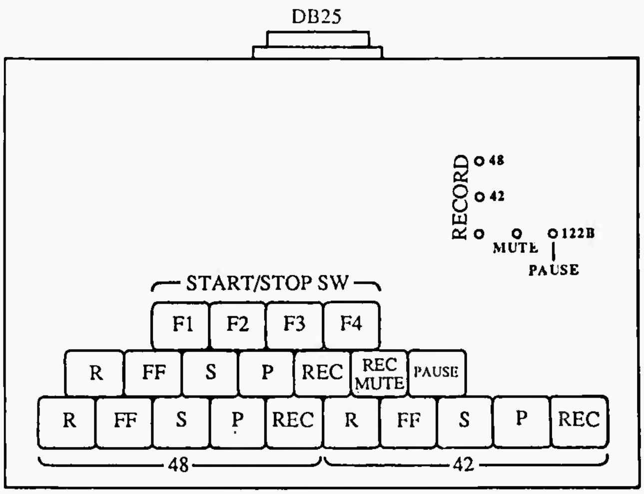


Figure 3. Control room.

Figure 4. Stuart Hollinger (left) and Wes Pacanas (right) in their Studio room A.

Because we are a new business offering specialized services, we felt it was important to be located in an easy-access, high visibility area. We found the perfect spot in the Lihue Shopping Center; our front door faces the main intersection in town.

The structure itself left a bit to be desired, but we managed to improvise and be flexible. The building is a hollow block oval shaped design. One good point was the room had no parallel walls and it was void of any existing partitions. This left us free to design the floor plan as we wanted.

Our reception room/lounge is 10-ft 6-in x 14-ft 6-in, the control room is 10-ft 6-in x 12-ft 0-in and the studio is 14 x 12. There's a 4 x 5 double plate glass window that allows viewing between the control and studio rooms. The wall partitions are 6-in thick with a sound air trap and a 3-in fiber glass coating inside. We used approximately 12 sheets of 4-ft x 4-ft Sonex, cut and placed in strategic areas.

One problem we ran into with the hollow block was sound vibration. Three sides of the whole structure is solid, but the back wall of the studio room is single hollow block, separating us from the office next door. Due to room size we did not want to build a parallel wall so we installed particle board directly to the block wall and added a layer of 4-in Sonex. This effectively nullified the unwanted vibrations.

Since our opening in August of 1986, our response has been tremendous. As of January of this year we are expanding into a 1000 sqaure foot structure. The building and use of our present studio has been a wonderful learning experience. The knowledge that we have gained is put to good use in the construction of our new studio.

Fortunately, our equipment was chosen with just such an expansion in mind. Future growth was one of the main factors in equipment selection. Instead of becoming obsolete, as the studio grows, the equipment grows with it. Because our tape machines have a synchronizer interface, our system can expand into 16-tracks by syncing two eight tracks via SMPTE time code. Or we could sync to video with a synchronizer because it is modular. The Soundtracs mixer can also be expanded by adding additional subgroup modules and input modules as needed. The patch bay has been pre-wired to accommodate 16-tracks.

THE EQUIPMENT

Our system consists of a Tascam 48 8-track connected to a Soundtracs T series mixer. Effects are patched in through a custom-made patch bay...built from surplus phone company patch bays. The patch bay is made up of modified phone company patch bays in single row with twenty jacks. They were then connected in pairs and wired together to have normalled connections where necessary. All jacks are 3-conductor 1/4-in with matching 1/4-in phone company plugs that are all brass with 2-in. long handles. Critical audio connections are connected through the patch bay with balanced lines. All cabling throughout the patch bay system is from phone company surplus and is extremely quiet and very low capacitance.

Reference monitors are JBL 4401 and Radio Shack Minimus 17 speakers. A Tascam 42NB half track handles the mixdown chores. We also have a Tascam 122B cassette machine. Controlling all tape machines is a custom built remote control unit. The unit contains all transport functions for the Tascam 48, 42 and 122 cassette. It also includes status indicator lights for the decks (ie: Rec., Pause). Wired in are switches for remote start/stop for drum machines, sequencers and other equipment that needs remote start/stop functions. The unit is made of a computer keyboard wired to a DB 25 connector.

Outboard equipment at Splash consists of DOD 6400 Digital Reverb, DOD 3.6 Digital Delay, Fostex Compressor/Noise Gate and Fostex Graphic EQ.

Although Splash is an 8-track analog studio, a total of 24 tracks can be recorded by syncing the Korg SQD-1 Sequencer to tape; 8 tracks acoustic sources and 16 tracks digital sources. To index all track information, data is kept on file by a Tandy computer with a track sheet program that can also remember mixdown settings and graphically display them.

The studio and control room are linked together through MIDI, so MIDI instruments can be controlled from anywhere throughout the studio. Also in use is a Kramer Pitchrider Guitar to MIDI interface used for sequencer programming or unique "guitar solos."

Other equipment includes: EMU Drumulator, Korg Polysix, Poly 800, Sci Max Keyboards, Electric guitars...Takamine TX 1000B, Musicman Bass, Acoustic guitars...Takamine F-340, Gibson L-7 and Martin 018.

Although Splash is designed and equipped as a high-tech, state-of-the-art facility, our business still functions on old-fashioned courtesy and personal customer service!

For mainland clientele looking for a quiet, beautiful place to remove themselves and put some basic tracks down, Kauai is a hard place to beat. It is the least developed of the major Hawaiian Islands. Kauai is still graced by miles of desolate white sand beaches and secret tropical waterfalls. Of course, if you miss the bright city lights, Honolulu is only a twenty minute jet flight away. Splash will assist in finding proper accommodations and offers a full-service recording studio. If equipment is needed that we don't have, we'll find it! Arrangements can be made to fit any budget. Why go to the Bahamas or the West Indies...keep it in America!

For local resident musicians, we offer assistance in duplication, distribution and marketing of their finished project. It's one thing to come in the studio and lay down some great tracks, but where's it going from there? We want to see musicians produce themselves and get their product out to the public. To promising musicians without the available capital to produce themselves, we offer a no-strings attached co-production plan. When our customers do well, Splash does well also! A happy, successful customer will return for more.

No matter how big Splash grows in the future, our ultimate goal will be to remain accessible and affordable. Not everyone's budget is geared for 16- and 24-track studios. Our idea of the perfect studio is one that can accommodate anyone's recording desires.

Here at Splash the motivating factor is...we enjoy what we're doing! ■

AUDIO-VIDEO COMPUTER BULLETIN BOARD NOW ON LINE

● Recently, a survey was completed on computer usage trends within the sound and video industries. The resulting tally indicated that many people are either just starting to learn about personal computers and communications via modem or are interested in the medium as a means of interacting with others through electronic mail (E-MAIL).

A second survey found that those who use computers on a regular basis and communicate with services which charge by the hour are, in many instances, not receiving their money's worth. This stemmed mostly from the fact that many of the pay-per-use services offer a scope much larger than special-interest users are willing to subscribe to.

In response to the needs of users in sound, video and related areas of the industry, a non-profit and simple-to-use bulletin board service, named A-V SYNC, has been formed and is now on line. The bulletin-board service (BBS) is dedicated to the free exchange of ideas, special interests, and discussions among professionals in the audio, video, and production-oriented areas of the industry.

The system features multiple lines which may be accessed through either standard dialing or various networks available to users of personal computers.

Reaction within the industry has already been so positive and diverse that several areas had to be addressed. They are designed to cover most special interests in the industry by groups of conferences or forums. The conferences are divided into the following groups:

- Audio for Video
- Television Sound
- Production Q and A – practicum
- New Technology (including digital)
- Manufacturer's Forum (new products and feedback)
- Trends (open discussion of systems and operations)
- System Files conference (special software files and text)

Where applicable, there will be new or modified files available for users of software-based mixing and editing systems along with updates and related text files as supplied by various manufacturers and programmers.

For further information, contact **AV-SYNC** at their 24-hour answering machine at **404 438-5858**. The system modem number direct is **404 320-6202. AV-SYNC** is also available on **PC Pursuit.** Or you can write to **AV-SYNC, PO Box 49567, Atlanta, GA 30359.**

JOHN BARILLA

Hot Tips for the Smaller Studio

"It's only an eight-track recording..." "So what do you want from a four-track demo!" "Well, it's sixteen-track, but you know it's narrow gauge stuff..."

Have you ever made these kinds of excuses? Do you sometimes feel it's necessary to issue this kind of disclaimer before playing your product for some important person? If so, this article may offer you some insights on how to get more out of your small recording setup.

The premise is simple: every system has its limits—the point beyond which it is absolutely impossible to get a better sound. But suppose you really *haven't* squeezed out every last drop of moxy from your equipment? Then the current limit to your system is *you*!

Let's face it folks, engineering chops can make a big difference. The care with which you cut your tracks, and even the way you strap various pieces of gear together can radically affect the sound. In an age where virtually all small format multi-track recorders (with built-in noise reduction) have a signal-to-noise ratio in excess of the vinyl LP, you should be challenged to compare your work with a commercially released record.

The fact is, some very successful records have been made on small format gear. Records bearing the semi-pro "stigma" of Tascam, Fostex and Akai are occasionally slipping into the charts alongside the likes of Sony, Otari and Studer. While the differences between the two classes of equipment are dramatically audible in clinical situations, by the time it reaches the consumer, few people seem to notice. Even more prevalent is the use of small format for jingle production, television and film sound tracks. The technical "apartheid" appears to be softening quite a bit, so seize the opportunity and learn what you need to get broadcast quality from your studio.

THE DEVIL AND THE DEEP BLUE SEA

One is distortion and the other is noise, and as engineer you are caught in between all the time. As the operator of a narrow gauge studio, one attitude you simply cannot afford to have is complacency. If you want playback to sound anything like the promised specs, you are going to have to work for it. There is simply no way to play it safe. Learn to recognize distortion by experimentally pushing the limit of your record level until you hear the subtle beginnings of infidelity. The closer you are to distortion, the further away you are from the system's inherent noise. Still, you need to be extremely careful at first, because built-in noise reduction systems tend to saturate tape with extra high frequency energy. In short, you've really got to experiment until you are sure you've "maxxed-out" your system. Once you discover (and note) the limits for a particular class of sounds, subsequent recordings become easier. While watching the meters is a valuable tool, truth is, the ear is the final judge. If you can learn to finagle a few extra decibels of sound on tape (without, of course, the slightest hint of distortion), you are beginning to practice the *art* of engineering.

John Barilla operates a private-use facility in New York—based around an AKAI-1212.

AS THE CROW FLIES

That, is said to be the shortest distance between two points. Likewise should the wiring scheme be in your home studio. To keep costs down, manufacturers have opted for the "unbalanced" system of interconnections for what they consider to be "consumer" or "semi-pro" products. Despite what anyone may tell you, this is *not* inherently an inferior system to the pro standard of "balanced" lines; it just has an absolute commandment for optimal performance: Thou shalt keep thy cable paths short!

So examine your conscience! Have you been using a twenty-foot guitar cord to make a two-foot run? How many places have you committed a similar sin? If you have a rat's nest of wires connecting your equipment together, repent now! Get out your soldering iron and make your own cables—cut exactly to the proper length—and your system shall be healed. While you're at it, make sure to reroute all power cords away from your audio cables. If that's not totally possible (and usually it isn't), then make sure they cross at right angles where magnetic induction will cause the least interference.

YOU GOTTA HAVE CONNECTIONS

One of the great joys of a professional studio is the flexibility of patching. Anything can be connected to anything. This is part of the creative process that encourages the development of new sounds. If you already have a patch bay on line, enough said; you know the joy of patching. But one feature that most smaller studios neglect is a section of "mults." Mults are simply parallel connections that allow the output of a signal to be split into several other identical outputs. This lets you process one sound several different ways *simultaneously* rather than stringing them along in series. (After processing, these parallel paths can be re-integrated into the main signal path through spare channels on your console—if you have any channels to spare. If not, an auxiliary mixer can be used to sum the sounds together first—even the simplest "micro-mixer" can do a handy job here).

Mults are easy to make. Simply allocate some unused points on your patch bay and tie them all together in the rear—high to high, low to low, and/or ground to ground. Then anything you plug into one point will appear at all the others. Having mults will greatly increase your options in signal processing and alleviate the noise buildup that sometimes occurs when devices are chained together in series.

RING IT OUT AND SHAKE IT DOWN

Sounds like a good title for a dance tune, doesn't it? But it's an even better idea for the sound of your studio if you first "ring it out." By that I mean turning everything up to a maximum level (with no signal input) and listening to the inherent "garbage" in your system. That's right: channel faders up, master faders up, monitor control up and every normally attached piece of gear turned on in the nominal operating condition. What do you hear? Is it hiss? Hum? A little of each? A lot of both?

Every system (even state-of-the-art) has a certain amount of evident noise when everything is at full throttle. What is the best you can expect from *your* system? To find out, begin to "shake it down." Start from the periphery of your system (outboard gear) and turn pieces off, then on, then off, then on—one piece at a time and observe the noise level. Do the same with channels, faders and switches at your recording console. Does the noise level drop radically when one piece of gear is turned off? If so, focus in on that piece and determine why. (Don't freak out and send it out to the repair shop. Chances are it's something like a bad audio connection, improper grounding or faulty operating levels: things that can be easily remedied at home). On the other hand, if your overall noise level changes only slightly with the addition or deletion of the element in question, then you have arrived at an understanding of the inherent noise of your system. (This is admittedly a Neanderthal method for getting a handle on the noise floor of your system. If a full-time "tech" is not in your budget this year, you will find it a helpful qualitative technique).

Now reduce all faders (and monitor controls) to their normal operating level. The system should sound relatively quiet by comparison. Now that you know how the system sounds when everything is "pedal-to-the-metal," you have a standard by which you can judge any future degradation of the sound. So if you ever get that uneasy notion that the system just isn't sounding right, you know what to do: ring it out and shake it down!

FRIENDS YOU CAN RELY ON

The compressor and the expander are the smaller studio engineer's two best friends. When properly utilized, they can help you get that clean, "in your face" kind of sound heard on most of today's records. The settings you choose are, of course, a matter of individual artistic expression, but technically they serve the recordist in an even more important way. First of all, the compressor/limiter can help you put a hotter overall level on tape by controlling the dynamics of the input signal. Considering that your tape recorder has a little handicap in the signal-to-noise catagory (relative to "professional" format machines), too much dynamics simply means that part of your sound is going to reside dangerously close to the noise level of your tape. So why not record your material closer to the "sweet spot" and limit or compress almost everything that hits tape? (I can hear the audio purists snooting at me already, but no recordist would complain about tracks that sounded cleaner and more like a record than a tape, would they?)

While we're talking of cleanliness, why don't you run most of your tracks through an expander as well. If you mess with it long enough and learn how to optimize the settings for the various kinds of input signals, it will make a radical improvement in the sound of a small format studio. The expander/gate can help shape the dynamics of the sound even to the extent of shutting a channel off when no meaningful signal is present. Here a little, there a little, those dBs of silence really add up. This can really help your system live up it's fullest potential. Units that house two channels of selectable compression/limiting, expansion/gating are all over the marketplace. (Personally, I'm a fan of the Symetrix 522 for a versatile, cost-effective unit, but there are several other candidates). So get one and cultivate the friendship!

EXTEND YOUR HIGHS

Let's face it. There are some deficits in the high-end response of small format multi-tracks. Spinning tape at slower speeds, recording at lower levels and narrow track width all conspire to a high frequency roll-off. Most of these rigs look pretty flat out to 12KHz or so, but who knows what happens at 16 or 20 KHz. (You may wonder who cares what happens up that high since most of us can't really "hear" much up there anyway. But hear it or not, when the stuff is missing, it shows!) If you wait until the mix to notice this deficiency and try to fix it with EQ, you will also be boosting all the high frequency hiss that's coming off the tape. So it's a good idea to add a little (Remember I said "a little") of that good high stuff on most every track that goes down. (The notable exception might be bass instruments which generally don't contain harmonics up that high). If upon mixdown the track sounds a mite bright, you can always roll it off, and as a result some of the residual tape hiss will be diminished as well. This is a great technique, but needs to be implemented scientifically. Experiment until you find the right amount of pre-emphasis for your system. Once you do, you will have succeeded in pushing back another limit of small format recording.

LITTLE THINGS MEAN A LOT

This is the great commandment for small recordists: *Cultivate the habit of meticulousness in all things.* This is above all an attitude- a mindset- which has to be developed in order to get broadcast quality product out of a small format studio. We're talking "seat-of-the-pants" rather than "state-of-the-art" engineering, so the bottom line is this: in order for you to get product out of your studio that sounds *nearly* as good as a professional shop, you have to be *twice* as good an engineer. (When I say "twice as good," I don't mean twice as technically astute, I mean twice as diligent!)

Little things really do mean a lot. Like cleaning your tape heads. (When your track width is less than 1/16th of an inch- in some cases 1/32nd of an inch, you should realize that a speck of oxide can seem like a boulder to your tape recorder.) Like keeping channels turned off when not in use. Like making sure all "send" levels from your mixer are as hefty as possible, so that your "returns"- and any consequent noise- can be kept to a minimum. Essentially, it's like being obsessed with perfection, and crafting every track as a virtuoso until each nuance of your system is known and mastered!

So remember this: Hit records, commercials and sound tracks are being recorded on equipment like yours. It is possible... ■

Meat and Potatoes

COREY DAVIDSON

We bring you a chronology of the construction of a world-class recording studio from its beginnings, building, and the completion of its site in Glen Cove , Long Island (NY)

Cove City Sound Studios co-owner and chief engineer Clay Hutchinson states: "This fortress of a building, erected in 1906 as *The Carpenters Meeting Hall*, had to be totally demolished and reduced to a shell in order to begin a new construction that would accommodate a unique recording facility."

The criteria for the guidelines of construction was based on instincts...instincts that were acquired from years and years of recording, performing and engineering. Co-owner and producer Richie Cannata explained, "When I walked into this building for the first time, my attention was immediately drawn to the height of the ceilings. The building was essentially one huge open space with a ceiling that ranged from 32 to 36 feet." (*Figure 1*) There are studios going up all over the place. One might ask why this location is different from all the others, and what led Clay and Richie to believe that they would have any kind of an edge over other designs by starting from scratch.

Cannata answered, "We had the total space to get the edge on other facilities. Basically, there were no restrictions on our construction concepts because of the tremendous available space." Clay's experience in building and designing studios was an edge as well. He was the first to have an automated 24-track studio on Long Island. "Clay and I pioneered a studio that is vast in ambient space. Every studio on Long Island has an 8 to 10 foot ceiling... if that." Other considerations for Hutchinson and Cannata were accessibility from other locations, competitive factors such as equipment and available space as so many New York City based studios have. "This studio is right off the L.I. Expressway. This not only facilitates our business traffic today, but when we were under construction, we had large, heavy equipment coming in and out of here. Our location was an advantage in the construction stages," remarked Cannata.

There are two basic philosophies that coexist here at CCSS. One is the unique approach in studio design and the other is the concept of "Meat and Potatoes." Cannata explained, "All the records that I made with Billy Joel as well as many others have been recorded on Neve and Studer equipment. These two brands in combination have established themselves as real workhorses, the backbone or as we say here, the "Meat and Potatoes." We're using the Neve 8068. Clay was flying all over the United States looking for, specifically, an 8068 console. Two years ago, when Clay was doing his shopping and searching for the console, we found that people were hoarding their Neves. Many of those people were SSL candidates and hadn't received their new consoles yet. Some were actually torn as to what to do and which way to go... Neve or SSL. A very musical board, which Clay will explain later, and a great analog machine, together, document the kind of sound that we feel is a true marriage of acoustics and electronics." The composite photo of the Cove City studio and control room exemplifies what Clay and Richie often try to explain to others. Clay said, "The cover shot is particularly interesting in that it predicts where recording is headed. It's not the obliteration of natural acoustics or the onslaught of total technology. It always will be both, acoustics and technology, because we are talking about quality sounds such as sampled sounds. The great sampled sounds are often captured with fine microphones and a big room." Richie added, "Once you've got the great sounds that you want, like horns, drums or vocals, then you can go nuts in the mixing and processing, but you've got to start with the fundamentals of what good sounds are made of and then the ability to control those sounds." The term "musical" is often used in conjunction with electronics. The notion that a console can be "musical" certainly gives us food for thought. Clay remarked, "Technically, when talking about the Neve's musicality, we refer to the wide bandwidth of response as it pertains to E.Q. In the cases of the SSL, Harrison and Trident consoles, there are much sharper peaks that occur when boosting a given center frequency. For instance, if you're boosting 4k, you will see a sudden 6db rise at 3900Hz. Likewise at 4100Hz, the curve shoots down 6dB. This has been the recent trend in console E.Q. design. The Neve has a broader, less sharp treatment of frequencies at a specific center frequency. This inherent Neve characteristic causes the engineer/producer to boost less. The answer might be to take out certain frequencies as opposed to boosting. Subtractive equalization is a much less understood technique and I have spent many years exploring the benefits of this later approach. The console has a certain kind of E.Q. that I explained as musical, but we've got outboard E.Q.s, like Pultec and Orban, that can facilitate the other types of E.Q. sloping, such as the pin-head, or as I often like to refer to it, 'the razor-edge effects.'

Figure 1.The Glen Cove Carpenters Meeting Hall before renovation

Figure 2. The console being hoisted to the second floor.

Figure 4. The framing and pouring of concrete for the control room's console platform.

The isolation between stages within a console is a major factor in maintaining headroom and cleanliness of signal. Clay declared, "One of the Neve's greatest assets is that there are transformers at the various stages of the console and the separate stages of the signal path are isolated due to transformer action."

Construction of the studio's spatial arrangement was the greatest undertaking. Many decisions had to be made, and some of those decisions could not be based on textbook formulas. Richie remarked, "When the space here was yet undeveloped, Clay had put string up to outline what he wanted as a control room. He set up a chair in a position that emulated a mixing perspective, and got up many times from the chair to readjust the strings." Clay described, "I had gone through numerous texts to make sure that my instinctual designs were not grossly out of proportion." When the control room was completed, Al Feirstein from Acoustilog Inc. was called in to analyze the room. Richie recalled, "When Al first showed up, he looked around and said, 'I think you're going to have problems here... big problems, big problems.' He proceeded to set up the analyzers, took the measurements and said, 'I can't believe it! You guys lucked out!' That's how our first analysis session went." Clay added, "Upon final analysis of the control room, some resonance of the wood in the room became apparent. The only correction necessary was a slat/space design mid range trap that was installed directly above the console. I am extremely proud of how flat the control room is today. We have achieved the goal of revealing the clinical curve of the UREI 813 in an ideal environment."

Figure 3. Early control-room construction. Note the soffit and speaker cavities.

Figure 5. The steel ring for the parabolic reflector. Its approximate weight is 600 lbs.

Figure 6. Welding the ring's support structure in studio A.

Figure 8. Lifting the "I" beam supports to their positions for welding.

An interesting harmony exists between Clay's designs and the acoustical properties that were employed in order to reinforce the monitors without coloring. Clay mentioned, "During the initial phases of construction, I had the bare rooms analyzed. With every major change and addition of materials, subsequent measurements were taken. Acoustically speaking, I always had a good idea as to where I stood, however, as construction progressed, things became a little more predictable."

CLAY'S DESCRIPTIONS

The Cove City studio construction utilized four construction crews successively. "I had a hard time communicating with the first couple of crews. It wasn't important to me that I had people that knew how to build studios. What was more important was that I had construction people that would listen to what I had to say and what I told them to do." Using non-studio oriented builders was a very effective cost-cutting measure. "I would be sure this time to have no parallel surfaces in any of the rooms and as few traps as possible."

The building has three floors. The studio (Studio A) encompasses the second and third level. The available space for the complex is approximately 4500 sq. ft. There is 2500 sq. ft. of floor space on the first floor and 2000 sq. ft. on the second floor (offices, tape library, bedroom, extra bathroom).

Figure 7. Locating the center position of the studio for the ring placement in the ceiling. Note the isolation booth in the background.

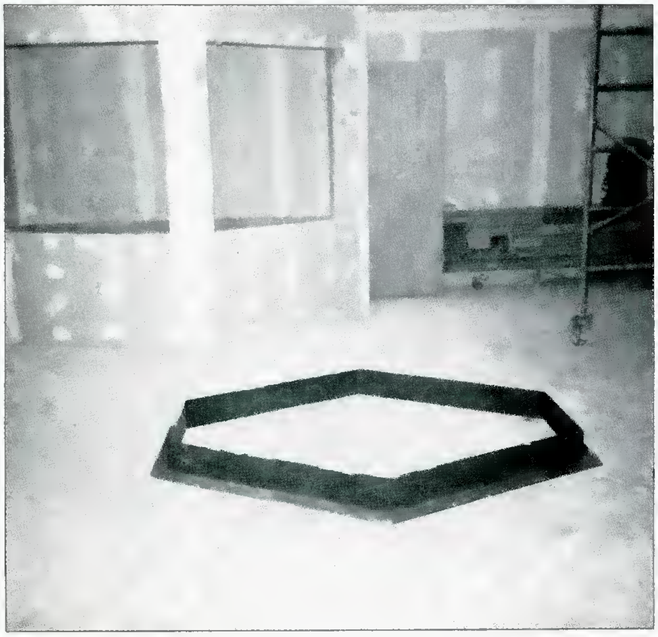


Figure 9. Raising the tripled 2x12s forming the ring's spider legs. The workers are standing on the isolation booth.

Figure 10. Central convergence, at the ring, of the spider legs.

Figure 12. The nearly completed control room. These are cedar walls and walnut parquet flooring.

"The console was hoisted in its crate (*Figure 2*) via forklift up and into the second floor lounge area where it remained during construction. The control room (*Figure 3*) was double sheet rocked, to my specs, with soffits and cavities for the speaker cabinets." (speakers had already been chosen). "The console platform framework (*Figure 4*) was constructed from 2 x 12-inch planks about one foot apart. A concrete pumper was brought in and pumped about 5 yards of concrete into the platform frame including the beveled ramp. Steel "I" beams were trucked in to support the hexagonal ring which was to be suspended from the ceiling in studio A. The steel hexagonal ring was of a tremendous weight (*Figure 5*). "I" beams to support the ring were necessary, and the welding of the ring support structure was conducted in studio A (*Figure 6*). "Due to the lack of right angle references, all measuring, cutting, welding and fitting had to be done on the fly." The hexagonal ring was laid on the floor in order to arrange placement in the center of the ceiling (*Figure 7*). Plumb lines were dropped from 24 feet up to accomplish centering. Steel "I" beams were hoisted (*Figure 8*) and welded to steel that was installed in the walls. These beams hold up the ring. Tripled 2 x 12-inches were raised (*Figure 9*) to form the spider-like legs radiating from the ring out to the walls and corners of studio A. The other spider legs are raised and mounted (*Figure 10*). Each leg radiating out from the hexagon meets the wall of the room at a different height. This staggering helps to enhance the uniform diffusion characteristics. Lumber is added to the hexagonal ring and the parabola begins to take form (*Figure 11*).

Figure 11. The basic structure of the parabolic reflector frame and radiating legs.

Figure 13. The rear, dead end, of the control room.

Studio A at Cove City Sound Studios.

Figure 14. Glen Cove Sound Studios' Grand Cafe renovation and reconstruction.

The parabolic reflector hangs eight feet down from the roof of the building. A catwalk between the roof and studio ceiling allows access in order to add and remove panels for the desired reflective qualities (room tuning)."The parabolic reflector established studio A as an amazing live drum room. Many of the world's best drum and percussion sounds are still recorded live. The priority in most modern music is rhythm. My studio reflects this priority. The decision to use a convex rather than concave dish was made three-quarters of the way into construction. By my analysis and preliminary measurements, the convex version would lend itself to better frequency diffusion properties for this space." The ceiling was then sheet rocked and insulated.

The control room (*Figure 12*) was finished in cedar walls and walnut parquet floors. Two panes of glass ; 14 x 5-ft., 5/8-inch thick (control room side), and 1/2-inch thick (studio side) were skewed and mounted. Pine planks (*Figure 13*) are installed at 45 degree angles about 1-inch apart to form the dead end of the LEDE control room. The two outside walls of the studio and control room were lined with lead. This lead lining and other necessary evils (2 x 6-inch instead of 2 x 4-inch, doubling 5/8-inch sheet rock, double insulation, staggered studding etc.) were absolutes in my designs."

"There are no right angles here, thus eliminating unwanted redundancies of reflected information. The parabolic reflector is the key to the sound here. Reverb time is controlled by diffusion. This diffusion allows for equal treatment of all frequencies in the spectrum."

The control room was designed around the speakers. "The design of the control room, when analyzed, reveals the clinical curve of the UREI 813s in an ideal environment. The room does not shade the color of the speakers whatsoever. There is no E.Q. to speak of other than a very slight roll-off on the bottom end because the UREIs are designed to have a 4-6dB boost at 125Hz. I chose to cut that a little bit. It helps to enhance a sensation of brightness that has become so desirable in modern recording techniques.

"Above the console, in the ceiling, is a 4-foot recessed area, hidden by baffling which acts as the mid-range trap. The control room is a LEDE type design. On the third floor, down a hallway, in a separate climate-controlled room, you will find the console power supplies, power amps and equalizers. The isolation permits a greater tolerance of temperature variation in the control room (a plus for clients who complain of freezing or boiling), not to mention a further reduction of noise. We installed an ionizer in control room A. It gives a fresh, crisp quality to the air. All air-conditioning ducting is oversized flex tubing, also eliminating noise. "

The studio contains two iso-booths, each one being of different dimensions. Subsequently, these rooms have varying sonic characteristics. Clay stated, "In the studio, there are actually three rooms. The booths have different volumes of space. The sound of plain dialogue (talking) through the same microphone sounds completely different from iso to iso. The difference in tambres is due to ceiling heights and volumes of space."

The concept of the *manageable room* has been realized at CCSS. The parabolic reflector is the key to the room's manageability. Reverb time is controlled by diffusion, and the diffusion enhances equal treatment of all frequencies in the spectrum. Richie declared, We can tune the room from *dead* to *live* and anywhere in between." It should be noted that the parabolic reflector is convex which serves as a *room activator*. Any sound waves coming in contact with the parabola are immediately thrown off the reflector and back into the room. This phenomenon is omnidirectional. The diffusion of sound is an efficient means of fully activating a space with sound. An analogy of this effect might be of ocean waves smashing onto rocks, scattering the water in all directions.

The CCSS MIDI room, headed by Ric Wake, has reinforced the acoustical strengths already present here. Clay said,"We believe that the computer/MIDI facilities here and the acoustics are equally important." Richie stated,"We look for diversification in the people we hire. Ric Wake has screened his programmers for technological and musical abilities. It's not enough to be a 'techno-tweaker' when there are so many musical aspects to cover. Our computer/synthesizer specialists are part musician, arranger and producer. When it's time to assemble an already existing piece of music, such as a previously recorded demo of an artist's material, it's a coffee break for Ric and his staff to commit it to MIDI."

Richie Cannata's input has taken an interesting form. Having travelled the world with Billy Joel, all his U.S. and worldwide tours from 1975-82, "I have seen the cream of the world's studios. I've seen utilitarian designs up through and including the unbelievably posh and elaborate 'fun' complexes." The ergonomics that Richie has brought to this studio are the result of a heightened awareness of so many different designs. "Comfortable yet undistracting concepts were employed, making it easy to spend a great deal of time here on the premises, while maintaining a super-low fatigue factor."

Slanted rolling bay outboard racks make for easy access to cabling and fast installations. In the control room, additional seating for clients exists (the producer's lounge) and is situated outside of the "critical sound zone." Kitchen and lounge facilities with chef-cooked meals are available and unusually convenient and parking is located just outside the front door.

Future plans include the utilization of additional space on the first floor. Clay explained, "In this space I intend to install a Neve V series, two digital machines, Urei 813s and an eclectic array of outboard gear. Adjacent to this space there will exist a small overdub room for last minute recordings. The intention is to create a control room environment. This will all serve as a mix room. Essentially that room will be a scaled down version of control room A's concepts."

Cove City Sound Studios has left no stone unturned. The incorporation of the MIDI room and computer interfacing has been a compliment to the exceptional designs present here. There is no doubt that personnel and experience provide the most service.

Clay concluded, "In closing, I would like to say that we are more confident than ever that CCSS will be turning out the kind of product that the industry and clients want and need. I would also like to thank db, to which I've subscribed to for over ten years now, for deeming this article a worthy one." ■

EQUIPMENT LIST

Neve 8068 32/16
Neve 8014 16/4
Soundcraft 400B 24/4
2-Studer A-80 w/Lynx 48-track
2-B-67 1/4-inch 2-track
Ampex AG-440-C 2-track
Studer A-710
2-Nakamichi MR-1, BX-100
AMS Reverb
Lexicon PCM-60, PCM-70, PCM-42
Eventide 949, 969
2-Orban Stereo EQs
Eventide 1745 M DDL
3-dbx 160, 160X
4-UREI LN 1176 N
Alembic Tube Preamp.
Kepex II
dbx 166
Yamaha Rev 7
Neve Comp/Lim 332264
Aphex Compellor
6-AKG 4144-U-87, 24-57, 6-421, 2-441, 2-Tube U-47
2-RE-20, 4-RE-15
2-AKG-D-12, 4-AKG-452
UREI 813-B
Yamaha NS-10M
Altec 640 E
White 1/6 octave room EQ
Auratones
McIntosh 2255, 2155
Crown D-150, DC-300A
UREI 6500
AKG 240, 140 headphones

MIDI ROOM

Yamaha QX-1
Emulator 11 +
Juno 106, 60
SP-12
Mini-Moog
2-Yamaha DX-7
Memory Moog

Recording Techniques

BRUCE BARTLETT

Tape Recording, Part 2

• *(This is the second and final part of this column which was begun in the last issue.)*

Cleaning The Tape Path. Oxide shed from the tape accumulates on the recorder heads. This layer of deposits separates the tape from the heads, causing high-frequency loss and drop-outs. In addition, buildup of oxide on the tape guides, capstan, and pinch roller can cause flutter. So it's very important to clean the entire tape path frequently.

Use the cleaning agent recommended by the tape recorder manufacturer. Denatured alcohol (from hardware stores or drugstores) and a dense-packed cotton swab are often used. Note: Some manufacturers recommend using rubber cleaner rather than alcohol for rubber parts to prevent swelling or cracking. Clean your machines after every eight hours of use, before alignment, and before every recording session. Allow the

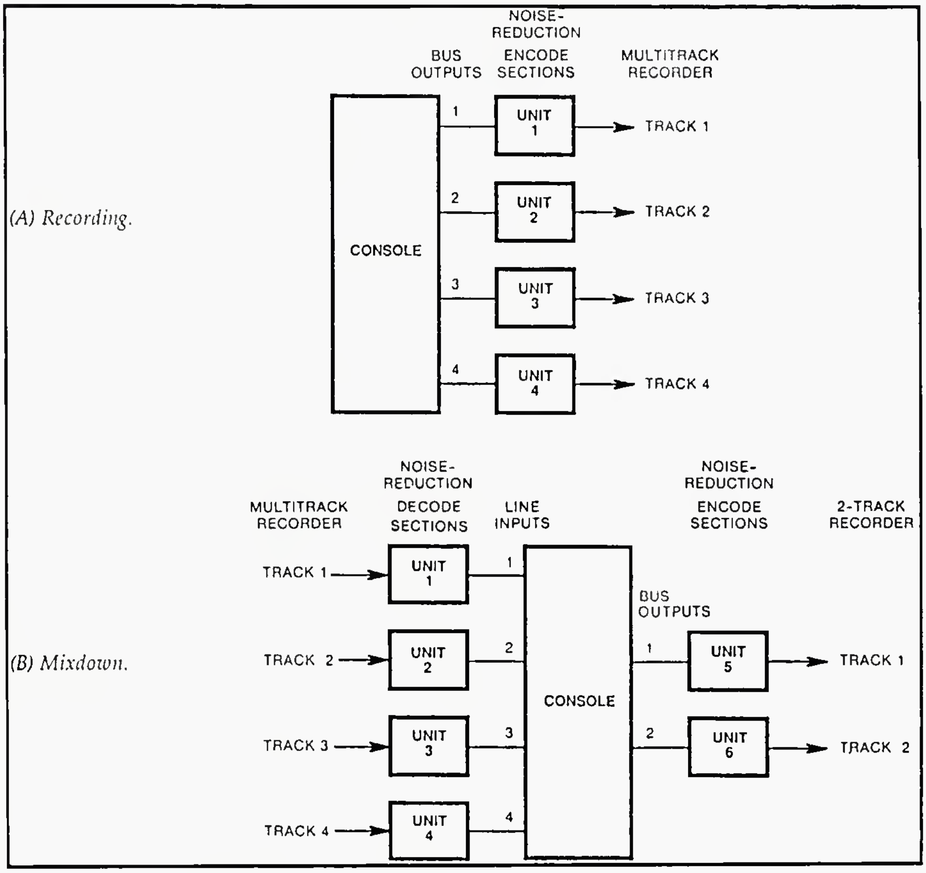


Figure 1. Noise reduction applied to multi-track tape and to two-track master tape.

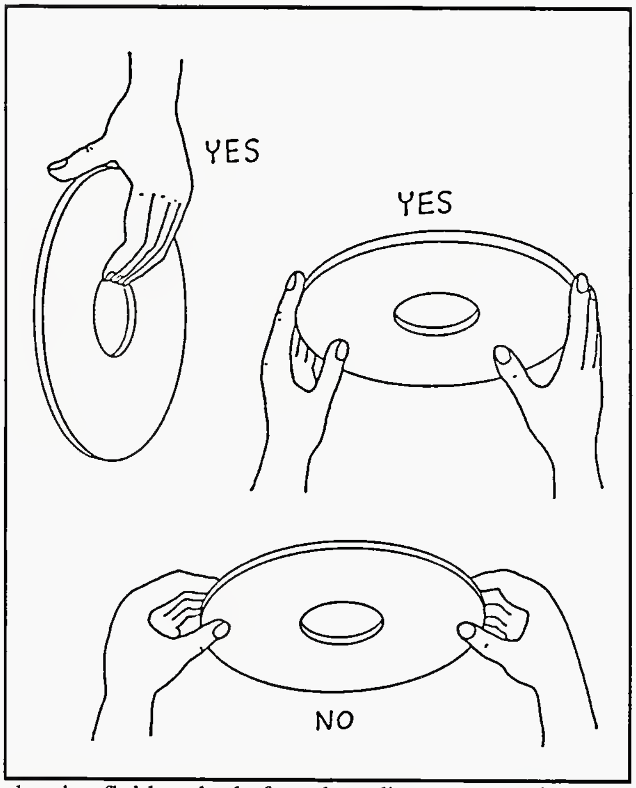


Figure 2. Handling tape reels.

cleaning fluid to dry before threading tape.

Demagnetizing the Tape Path. Tape heads and tape guides can accumulate residual magnetism which can partially erase high frequencies, add tape hiss, and cause clicks at splices. This residual magnetism can be eliminated with a tape-head demagnetizer or de-gausser, available from any sound system dealer.

Essentially an electromagnet with a probe tip, the demagnetizer produces a 60-Hz oscillating magnetic field. By touching the probe tip to the heads and tape guides, you magnetize them; by slowly pulling the tip away, you diminish the induced magnetization until no magnetic field is left. Generally, only the gapped types are strong enough to be effective.

Cover the probe tip with electrical tape, if necessary, to avoid scratching the heads.

The technique of using a demagnetizer is critical. Proceed as follows:

1. Turn off the recorder.

2. Plug in the demagnetizer at least three feet from the machine.

3. Bring the demagnetizer slowly to the part to be demagnetized.

4. After touching the part with the probe tip, remove the demagnetizer slowly to three feet away so that the induced magnetic field gradually diminishes to zero. Touching the demagnetizer to a head and quickly removing it may magnetize the head worse than when you started.

5. Demagnetize each tape head and tape guide one at a time in this manner.

6. Turn off the demagnetizer only when it's at least a foot from the machine.

Demagnetize your machines after every eight hours of use and before playing an alignment tape. The same precautions about slow operation and three-foot turn-off apply to a bulk tape eraser as well.

Alignment. Alignment or calibration is the adjustment of tape-recorder circuitry and tape-head azimuth for optimum performance from the particular type of tape being used. It's a complicated procedure not recommended for beginners. Professional recording engineers align their machines periodically to ensure flat frequency response, maximum signal-to-noise ratio, and lowest distortion.

Some home and semi-pro recorders are not designed for easy alignment. The internal pots to be adjusted may not be easily accessible. In that case, the alignment is usually left alone, and you use the brand of tape for which the machine was adjusted.

To perform a complete alignment, you'll need a small screwdriver, an audio-frequency generator, and a standard playback alignment tape. Information about such tapes is available from Ampex, 2201 Lunt Ave., Elk Grove Village, IL 60007; Magnetic Reference Laboratory, 999 Commercial St., Palo Alto, CA 94303, and from various tape-recorder manufacturers.

Follow the tape-recorder instructions regarding calibration. Clean and demagnetize the tape heads before starting. Basically, you'll perform the following operations:

1. Using the alignment tape, play the 15 kHz tone and adjust playback-head azimuth for maximum output or for best phase match between channels (using an oscilloscope).

2. Adjust the high-frequency playback equalization (if any) to achieve the same output level at 700Hz and 10kHz. Or try for the flattest overall response if several tones are on the tape. Don't adjust the low-frequency equalization yet.

3. The magnetic field strength on tape (the fluxivity) is measured in nanowebers per meter (nWb/m). If

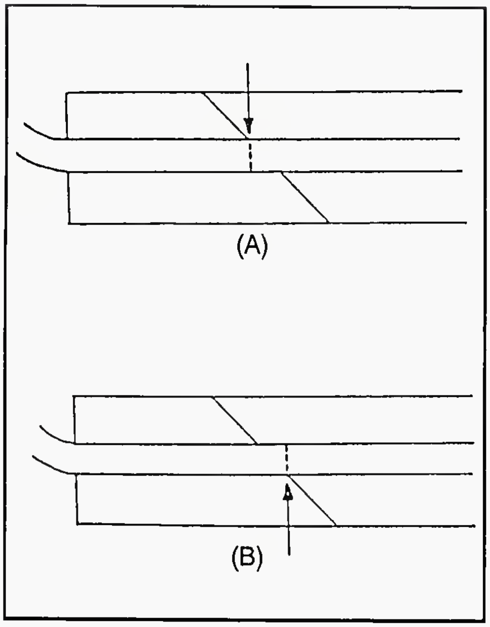


Figure 3. Aligning edit marks with the cutting slot on the editing block. At (A) the arrow points to the mark to be made at the beginning of a song. At (B) it points to the mark for the end of the song.

you're using an alignment tape that has a standard operating level of 185 nWb/m (old Ampex standard level), set the playback level to read -3VU or -6VU as recommended by the recording tape manufacturer. If you're using an elevated-level alignment tape that uses a standard operating level of 250nWb/m or 320nWb/m, set the playback level to read 0VU (or as recommended by the recording tape manufacturer and recorder manufacturer). Note: Do not touch the playback level for the rest of the calibration.

4. Thread on some blank tape of the desired brand.

5. Record a 15kHz tone and adjust record-head azimuth for maximum playback output, or for best phase match between channels. Note: Skip this step if your recorder combines the record and playback functions in a single head.

6. While recording a 1kHz tone, set the bias to achieve maximum playback level. Then go back to 10kHz, and turn up the bias past that point (overbias) until the output drops 1/2 to 1dB. Overbiasing reduces dropouts and modulation noise. Consult the tape manufacturer's directions for alternative overbias settings.

7. While recording tones of 10kHz, 100Hz, and 700Hz, adjust the high-frequency record equalization and low-frequency playback equalization (if any) to achieve the same playback output level at all frequencies. Or use many tones to achieve the flattest overall response. Record the tones at 0VU for 15 in./sec., -10VU for 7 1/2 in./sec., and -20VU for cassettes.

8. Feed a 1kHz tone at 0VU from the mixing console to the recorder. Record the tone. Set the record level so the recorder reads 0VU on playback.

9. Set the "record cal" or "meter cal" so that the meter reads 0VU on "input" or "source."

After calibration, your tape machine will operate as well as possible with the particular type of tape you're using. The playback signal should sound identical to the input signal (except for some added tape hiss).

Reducing Print-Through. Print-through is the transfer of a magnetic signal from one layer of tape to the next, causing an echo. If the echo follows the program, it is called post-echo. If the echo precedes the program, it is called pre-echo. Print-through is especially audible in recordings with many silent passages, such as narration. To minimize print-through:

- Use low-print tape.
- Demagnetize the tape path (because stray magnetic fields increase print-through).
- Use 1 1/2 mil tape (thinner tapes increase print-through).
- Use noise-reduction devices (discussed later in this article).
- Store tapes at temperatures under 80 degrees Fahrenheit, and don't leave tapes on a hot machine (because heat increases print-through).
- Rewind tapes in storage at least once a year to allow print-through to decay.
- Store tapes tail out. That is, after playing or recording a tape, leave it on the take-up reel. Rewinding a tape about 15 minutes before playing helps to reduce print-through that may have occurred during storage. (This measure becomes less effective as the storage time increases.) In addition, tail-out storage results mainly in post-echo, which is less audible than the pre-echo emphasized in tapes stored rewound.

Operating Precautions. Here are some operating tips for tape recorders that may prevent some accidents:

Don't put the machine in record mode until levels are set. If you record an extremely high-level high-frequency signal, the crosstalk within the head might erase other tracks.

Keep tape away from recorder heads when turning the machine on or off. Many recorders generate a field spike which may put a click on tape.

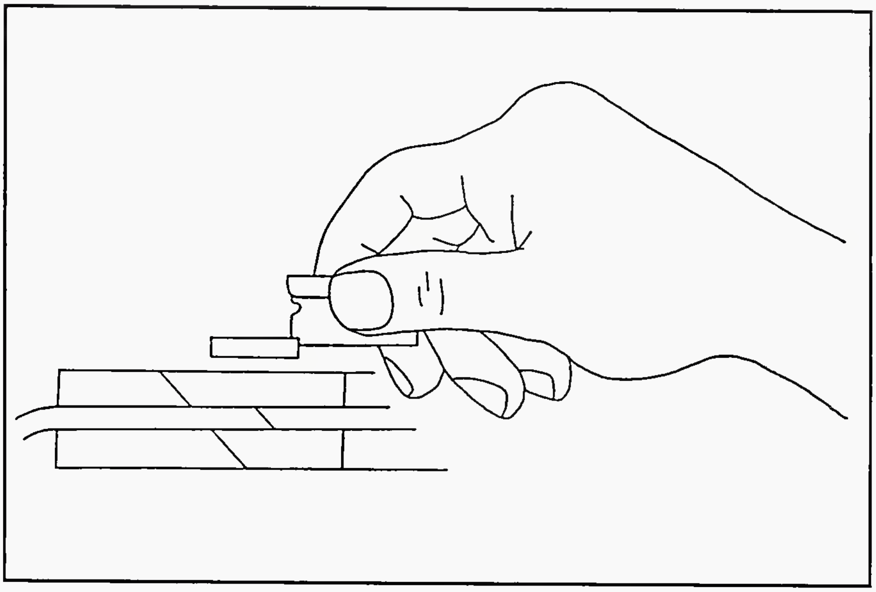
Keep de-gaussers and bulk tape erasers several feet from tapes you don't want to erase.

Before you start recording on a track, make sure you won't be erasing something you wanted to keep. Listen to the track first or refer to your track sheet.

Edge tracks of multitrack tapes are prone to drop-outs due to edge damage. Since drop-outs occur mostly at high frequencies, use the edge tracks only to record instruments with little high-frequency output (such as bass or kick drum).

Repeated passes of a recording past the heads may gradually erase high frequencies. You may want to make a copy of the multitrack tape (or a quick two-track mix) for musicians to practice overdubs with. Then go back to the original tape when the musicians are ready to record.

Figure 4. The best way to apply splicing tape.



Bouncing or ping-ponging is the process of mixing several tape tracks and recording the mix on a single open track. Then you can erase the original tracks to free them up for recording new instruments. Bouncing or copying tracks tends to lose high-frequency response, so try to limit bounced tracks to bass or mid-range instruments.

NOISE REDUCTION

The analog tape recorder adds undesirable tape hiss and print-through to the recorded signal, degrading its clarity. Tape hiss becomes especially audible during a multitrack mixdown because every track mixed in adds to the overall noise level. Noise increases 3 dB whenever the number of tracks in use doubles, assuming they are mixed at equal levels.

Fortunately, noise-reduction devices such as Dolby or dbx are available to reduce noise and print-through added by the tape recorder. These units do not remove noise in the original signal from the console.

One channel of noise reduction is needed per tape track. Noise-reduction units connect between the console output buses and the corresponding tape-track inputs, and also between the tape-track outputs and the console tape inputs, as shown *Figure 1*. Some multitrack recorders have built-in noise reduction.

These noise-reduction devices compress the signal during recording and expand it in a complementary fashion during playback. The compressor part of the circuit boosts the recorded level of quiet musical passages. The expander part works in a complementary way during playback, turning down the volume during quiet passages, thereby reducing noise added by the tape. During loud passages (when noise is masked by the program), the gain returns to normal.

A compressed tape is described as encoded; the expanded tape is called decoded. If an encoded tape is played without decoding, the dynamic range and frequency response are altered.

It's important that the encode and decode sections track each other. For example, a 10-dB level change at the input of the encode section should yield a 10-dB level change at the output of the decode section. Otherwise, dynamics sound unnatural.

The Two Major Noise-Reduction Systems: dbx And Dolby

dbx. With dbx noise reduction, the compression ratio is 2:1. That is, a program with a 90-dB dynamic range is compressed to 45dB, which is easily handled by a tape recorder with a 60-dB signal-to-noise ratio. Then during playback, the dynamic range is expanded back to the original 90 dB. Use of dbx improves signal-to-noise ratio by 30 dB and increases headroom by 10dB. The dbx circuit also includes pre-emphasis (treble boost) of 12dB during recording and complementary de-emphasis (treble cut) during playback to reduce modulation noise. dbx operates at all signal levels and across the entire audible spectrum.

Dolby. The Dolby A system divides the audible spectrum into four separate frequency bands which are compressed and expanded independently. In addition, Dolby operates only on quiet passages–those below about -10VU. High-level passages do not need noise reduction since the program masks the noise. This system reduces noise by 10dB below 5kHz and up to 15dB at 15kHz.

Dolby B, a lower-cost system for cassette decks, operates only at high frequencies to reduce tape hiss by up to 10dB. Dolby C works over a slightly wider range and reduces noise by up to 20dB.

Dolby SR (Spectral Recording) is the most effective Dolby system, reducing noise by more than 25dB over most of the audible spectrum. As a result, a recorder operated at 15 in./sec. with Dolby SR can have a maximum signal-to-noise ratio exceeding 105dB. See the previous issue of **db** for a fuller discussion of Dolby SR.

When using Dolby A or SR, you must record a calibration signal called a Dolby tone on tape before the regular program. This tone is generated by an oscillator in the Dolby unit. During playback, the level of the recorded Dolby tone is indicated on a Dolby meter. You set the Dolby input level so that the meter indication lines up with the Dolby-level mark on the meter. Then the expander circuitry will track the recording properly.

If the level is set improperly, the frequency response and dynamic range are slightly altered. Fortunately, there is room for error since these alterations occur in low-level signals and consequently are hard to hear.

Dolby Vs. dbx. Both Dolby and dbx have advantages and disadvantages. Compared to Dolby A, B and C, dbx provides more noise reduction and requires no calibration tone or careful level setting. On the other hand, dbx exaggerates drop-outs more that Dolby does. Dolbyized recordings are relatively free of noise "breathing," which is sometimes audible on a dbx-encoded tape as fuzziness accompanying bass- or bass-drum solos.

Using Noise Reduction. Dolby- and dbx-encoded tapes are not compatible with each other, and cannot be played properly without decoding through the appropriate unit. So if you plan to send your master tapes to another studio, check that the studio has the same type of noise reduction you want to use.

When using noise reduction, avoid saturating the tape while recording. Otherwise the attack transients may be altered during playback through the noise-reduction unit. If you are using dbx noise reduction, you can record at 3VU lower than normal for 3dB more headroom.

Encoded tapes should be copied by decoding first, then re-encoding while recording onto the second machine. Be sure to copy the Dolby tone if the master tape has one.

Matching The Mixer Meters And Recorder Meters With dbx In Use. It's common practice to set the mixer meters and recorder meters to track each other. That way you have to watch only the mixer meters while recording. Also, when the mixer and recorder are both peaking around 0 VU, they are operating at an optimum level for distortion and noise performance.

However, if dbx noise reduction is used, matching the meters becomes confusing because the encoded signal from the dbx is compressed. Let's first describe how to match meter readings without dbx:

1. If necessary, calibrate the recorder as described in earlier in this article.

2. If you don't have a test oscillator (tone generator) built into your mixing console, get an external oscillator. Set it to 1kHz and about -50dB level. Plug it into a mixer mic input and bring up the level to read 0VU on all the console meters. An alternative to a 1-kHz tone generator is a continuous sine-wave synthesizer note, "C" two octaves above middle C (1046.52Hz), or "B" (987.8Hz).

3. Send the 0-VU tone from the console to the recorder.

4. Put the recorder in record mode, and set the record level to obtain 0VU during playback on all the recorder meters.

If dbx is used, the procedure is as follows:

1. Turn on a 1kHz tone set to 0VU on all the console meters.

2. Put the recorder in record mode. With dbx switched OUT, set the record level to read 0VU during playback on track 1 of the multitrack recorder.

3. With dbx switched IN, adjust the record-trim pot on the dbx so that the recorder meter still reads 0VU. Don't touch the record level on the recorder. Note: Since dbx operation varies with frequency, you must use a 1kHz tone. If you don't have a 1kHz tone, just calibrate with the dbx switched out.

4. Repeat these steps for all the recorder tracks.

5. With the 2-track recorder's dbx switched OUT, feed a 0VU tone to buses 1 and 2 of your mixer (or the stereo mixdown buses), and set the record levels on the 2-track machine to read 0VU on playback. Repeat the above steps for dbx adjustments.

Now that the mixer and recorders are calibrated to match each other at 0VU, leave the recorder controls alone. Set levels with the mixer faders only.

The signal the dbx feeds to the tape recorder is compressed, so the recorder meters will wiggle less than the mixer meters. This makes it difficult to set recording levels by watching the mixer meters, so you'll have to watch the recorder meters instead when dbx is used.

TAPE HANDLING AND STORAGE

Careful handling and storage of tape reels is essential to avoid damaging the tape and the signals recorded on it.

If you examine a reel of used recording tape, you may see some edges or layers of tape sticking out of the tape pack. These edges can be crushed by pressure from the reel flanges, causing drop-outs and high-frequency loss. For this reason, never hold a reel of tape by squeezing the flanges together. Instead, hold the reel in one hand by putting your fingers in the hub and your thumb on the flange edges, as shown in *Figure 2*. Hold a reel in two hands with extended fingers on the flange edges.

To prevent edge damage during storage, leave tapes tail out after playing or recording to ensure a smooth tape pack. Repair or discard reels with a bent flange. Reels left out in the open can collect dust, so keep them in boxes. Store tape boxes vertically–not stacked. The preferred storage conditions are 60 to 75 degrees Fahrenheit, 35-to-50 percent relative humidity. Keep tapes away from magnetic fields such as those caused by speakers or telephones.

EDITING AND LEADERING

Editing is the cutting and rejoining of magnetic tape to delete unwanted material, to insert leader tape, or to rearrange material into the desired sequence.

Equipment And Preparation. Editing requires the following materials: demagnetized single-edge razor blades, a light-colored grease pencil, splicing tape, leader tape, and an editing block.

Leader tape is plastic or paper tape without an oxide coating, used for a spacer between takes (i.e., silence between recorded songs). Plastic leader is preferred over paper because paper can absorb humidity during storage, which causes it to warp. An editing block holds the tape during the splicing operation. It's easier to use than a tape splicer with hold-down tabs and allows more-precise cuts.

Before editing, wash your hands to avoid getting oily spots on the tape. Cut several 1-inch pieces of splicing tape and stick them on the edge of the tape deck or table. Also cut several sections of leader at the 45-degree slot in the editing block. A typical leader length between songs is four seconds, which is 60 inches long for 15 in./sec. or 30 inches long for 7 1/2 in./sec. While editing, try to hold the magnetic tape lightly by the edges.

Leadering. Suppose you've recorded a reel full of takes, and you want to remove the out-takes, count-offs, and noises between the good takes. You also want to insert leader between each song. This process, called leadering, can be done as follows:

First, wind several turns of leader onto an empty take-up reel and cut the leader at the 45-degree slot. Remove this take-up reel, put on an empty one, and play the tape to be edited.

Locate the beginning of the first song's best take. Stop the tape there. Put the machine in cue or edit mode so the tape presses against the heads. While monitoring the tape-recorder output, rock the tape back and forth over the heads by rotating both reels by hand; first rapidly, then more and more slowly. You'll hear the music slowed down and low in pitch. Find the exact point on tape where the song starts; that is, where it passes over the playback-head gap. Align the beginning sound with the gap. Using the grease pencil, mark the tape about 1/2 inch to the right of the gap (that is, at a point on tape just before the song starts).

Next, loosen or "dump" the tape by simultaneously rotating the supply reel counterclockwise and the take-up reel clockwise. Remove the tape from the tape path and press it into the splicing block, oxide side down. Align the mark with the 45-degree-angled slot, as in *Figure 3-A*. Slice through the tape with a razor blade, drawing the blade toward you. Don't use the 90-degree slot because such an abrupt cut can cause a pop noise at the splice.

Remove the unwanted tape to the right of the cut and put the take-up reel aside. Slide the cut end of the tape to the right of the editing-block slot, as in *Figure 4*. Put on the take-up reel containing the turns of leader tape, and insert the end of the leader into the right half of the block. Shove together the ends of the leader tape and magnetic tape so that they butt or touch together with no overlap.

Now, take a piece of splicing tape and stick a corner of it onto a hand-held razor blade. Align the splicing-tape piece parallel to the recording tape, as in Figure 4. Apply the piece over the cut onto the non-oxide side, and stick it down by rubbing with fingernails.

Slide the splice out of the block. Gently pop the tape out of the block by pulling up on the ends of the tape extending from both sides of the block. Twist the tape toward you while pulling. Check that there is no gap or overlap at the splice.

Now wind the tape onto the take-up reel and locate the ending of the first song. As it ends, turn up the monitors and listen for the point where the reverberent "tail" of the music fades into tape hiss. Stop the tape there and mark it lightly at the playback-head gap (at the center line of the head).

After pressing the tape into the block, cut the tape at the mark as in *Figure 3-B*. Remove the tape to the left of the cut. Splice the end of the first song to a four-second length of leader and again check the splice. Wind the first song and the leader onto the take-up reel and remove it. Then put on the take-up reel containing unwanted material you previously set aside. Splice it to the rest of the master tape.

Next, locate the beginning of the next good take you want in the program. Mark it and cut the tape. Put the reel containing the first song on the take-up spindle. Splice the tail end of the leader onto the beginning of the second song, then wind the second song onto the take-up reel.

You now have two songs joined by leader tape.

Repeat this process until all the good takes are joined by leader. Then you will have a reel of tape with several songs separated by white leader, which makes it easy to find the desired selection.

Joining Different Takes. What if you want to join the verse of Take 1 to the chorus of Take 2? You'll have to cut into both takes at the same point in the song, then join them. It takes practice to make an inaudible splice in this manner, but it's done every day in professional studios.

The two takes must match in tempo, balance, and level for the edit to be undetectable. To mask any clicks occurring at the splice, cut the tape just before a beat–say, at the beginning of a drum attack. An alternative is to cut into a silent pause. If you cut into a continuous sound such as a steady chord, a cymbal ring, or reverberation, the splice will be noticeable.

Let's run through the procedure. First play Take 1 and locate the point where you want Take 1 to stop and Take 2 to start–say, at the beginning of the chorus. Stop the tape there. Then put the recorder in cue or edit mode, rock the tape, and try to identify a beat or attack transient. At the point on tape where this beat just starts to cross the playback-head gap, mark the tape. Cut the tape at the mark and remove the take-up reel containing the verse of Take 1.

Next, put on an empty take-up reel, thread the master tape, and fast-wind to Take 2. Find the same spot in Take 2 that you marked in Take 1. Mark and cut it. Using splicing tape, join Take 2 (in the supply reel) to Take 1 (in the take-up reel you just set aside). Again, check that there is no gap and no overlap at the splice.

Play the spliced area to see if the edit is detectable. If not, congratulations! It should sound like a single take. If Take 2 comes in a little late, carefully remove the splice and cut out just a little tape surrounding the cut. Re-splice and listen again.

More Editing Tips. Suppose you've recorded most of a good take, but then the musicians make a mistake and stop playing. Rather than repeating the entire song, the musicians can start playing a little before the point where they stopped, and then finish the song. Then you can splice the two segments into a complete and perfect take. Editing is also useful for interjecting comments or sound effects in the middle of a song, or for making tape loops. You can even record a difficult mixdown in segments, then edit the segments together.

THE DIGITAL RECORDER

The type of recorder described earlier is an analog recorder. That is, the magnetic particles on tape are oriented in patterns analogous to the audio waveform. The drawbacks of this system are tape hiss, tape distortion, frequency-response errors, and speed variations (wow and flutter). Recently, digital recorders have been developed that eliminate these problems.

The digital recorder is beyond the scope of this article, but here is a brief overview:

During recording, a digital recorder measures or samples the voltage of the incoming waveform about 48,000 times a second. Numbers are assigned to these voltages. This process is called analog-to-digital (A/D) conversion. The machine then records the numbers on tape in binary code (1's and 0's). This code is accurate to 16 digits (bits). The binary numbers are in the form of a modulated square wave recorded at maximum level.

During playback, the binary numbers are read and converted back into the original sampled voltages–a process called digital-to-analog (D/A) conversion. Finally, the varying voltage levels are smoothed back into the original audio waveform by a low pass filter.

Since the digital playback head reads only two binary numbers, it is insensitive to tape hiss and tape distortion. Numbers are read into a buffer memory and read out at a constant rate, eliminating speed variations.

The resulting freedom from noise, distortion, print-through, wow, and flutter makes digital recordings sound extremely clean and clear. Unlike analog recordings, digital recordings can be copied with little or no degradation in quality. Lost data is restored by error-correction circuitry.

Currently, digital recorders are quite expensive. For stereo mastering, there's an alternative to a digital recorder: a digital-audio processor with Pulse Code Modulation (such as the Sony PCM-F1). This device digitally encodes the audio and records it onto a standard video cassette tape. You connect the analog signal to be recorded to the processor inputs, and connect the processor outputs (a modulated radio-frequency signal) to the inputs of any video cassette recorder. Then you set levels and start recording. During playback, the modulated video signal feeds the processor, and analog audio comes out.

A low-cost alternative is a video cassette recorder with the Beta Hi-Fi or VHS Hi-Fi sound system. Audio is recorded as a frequency-modulated RF signal on a video cassette. The performance approaches that of digital recording.

Thanks to digital recording, the original goal of tape recording has finally been achieved: to accurately store and reproduce our audio creations. ■

MARSHALL KING

A Review of Television Stereo

Ah, how simple it used to be! There was a day when comments made by anyone in the field of broadcasting or recording could be quickly and easily understood by the others in that general occupation. Somehow, the work of the studio technician was not at all that different from that of the transmitter engineer; in fact, they were often one and the same person doing both jobs in a single location. We might say the same for the technician operating the cutting lathe in a record plant and the sports mixer working in a coliseum; they spoke a common language that dealt with such problems as overmodulation, frequency loss and crosstalk. While these topics are still in the rudimentary vocabulary of most electronics people, ultra specific state-of-the-art concepts are being born almost daily which render general discourse with our fellows something akin to the jumbled jargon from the Tower Of Babel.

What video tape maintenance person, for example, has a ready grasp of the technical parameters the rock concert technician deals with each time he hits the road? What sound mixer, whose time is mainly spent setting up the orchestra and delivering the audio on major award shows, can rap with the transmitter engineer beset with the ongoing problems of stereo alignment for subcarriers? What Midi consultant can step in and create new laugh tracks for an absent McKenzie cartridge operator? Today we seem to speak different languages, punctuated with far too many acronyms which force us to carry around translation dictionaries in the form of trade journals.

How did this happen? Aren't we all in the same business? Perhaps we're not anymore. We are reminded of the sheltered little grandmother who said, "You were in World War Two? Then you must know my grandson, he was in World War Two."

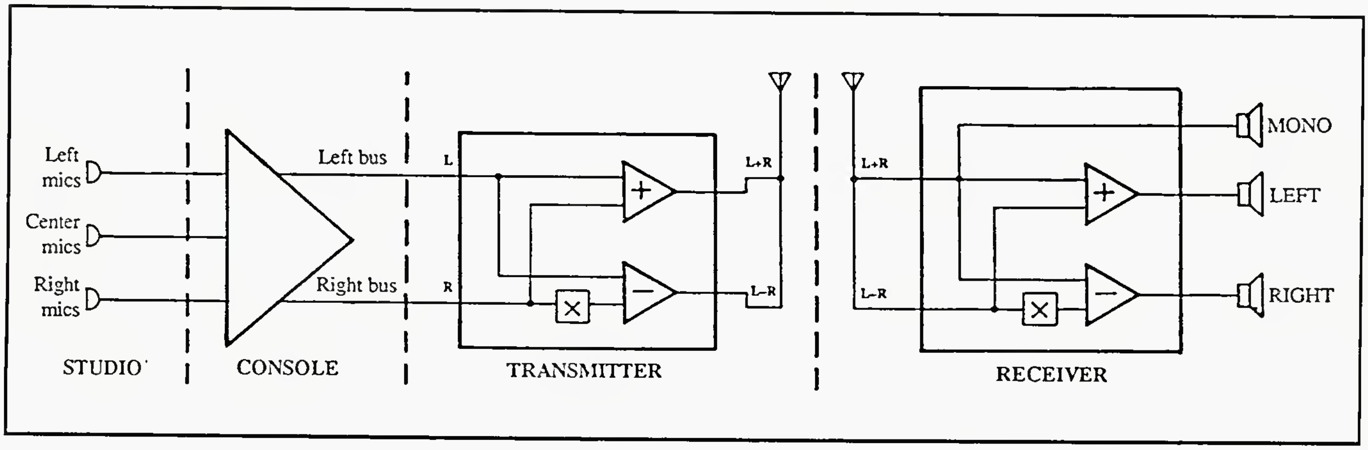
With this parable in mind, we present an overview of television stereo knowing that, even though we labor in the same field, we may be unaware of each other's work. While this may seem unimportant in view of today's specialization, the reality is that stereo is now a fact of life in television broadcasting and it can't help but touch our own work in some fashion. Since much of the information presented here is subjective, garnered from the experience and observations of many who were the first to get their feet wet, I hasten to say that suggestions, updates, criticism and corrections are most welcome.

Why stereo? There are probably two reasons. Firstly, since the sounds we deal with in everyday living come to us from many directions rather than from a point source, stereo reproductions have more authenticity than mono. Secondly, by bringing us into the scene (or rather, by wrapping the scene all around us), stereo gives us a feeling of participation that is denied when we're viewing and listening from a monophonic distance.

How to deal with stereo... how to record it, how to transmit it, how to receive and listen to it...demands that we first define what it is. There is a caution here; unlike phonographic records where sound is the only consideration, we are now dealing with pictures. The stereo concept should marry the two. Alas, it doesn't always do this, and hereby lies our first investigation.

Consider: When we listen to stereo music in our car, it's usually a fine and easy experience. The orchestra is fat and authoritative, the various sections coming from their proper place in the scheme of things. It does not matter if the brass is up front near the glove compartment, the drums somewhere back by the spare tire and the reeds in the vicinity of the dome light; their positions could be rotated for all we care. After all, since we'll never know where they were actually sitting during recording, it makes no difference where they seem to be located now, as long as a separation exists to give us a spatial effect.

Figure 1. The transmitter both adds and subtracts the left and right signals from the studio and sends them out as the sum (L + R) and the difference (L-R). From these two signals the receiver extracts what it needs to form mono, left, and rights channels.



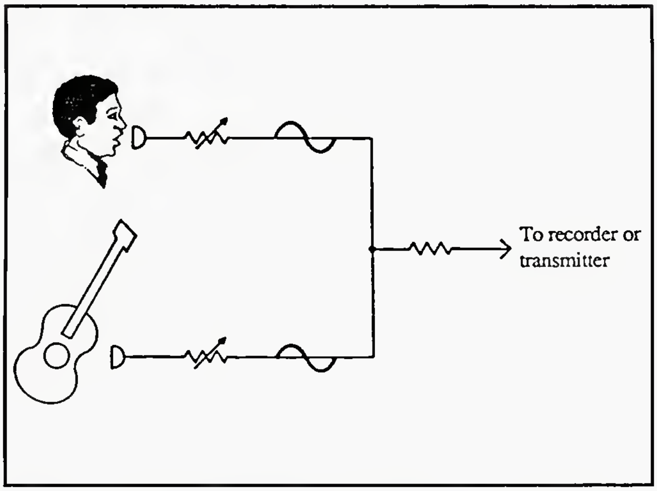


Figure 2. The sum is nothing more than the combining of any two or more signals channeled through circuits which are electrically in phase. Thus the term "Mixing".

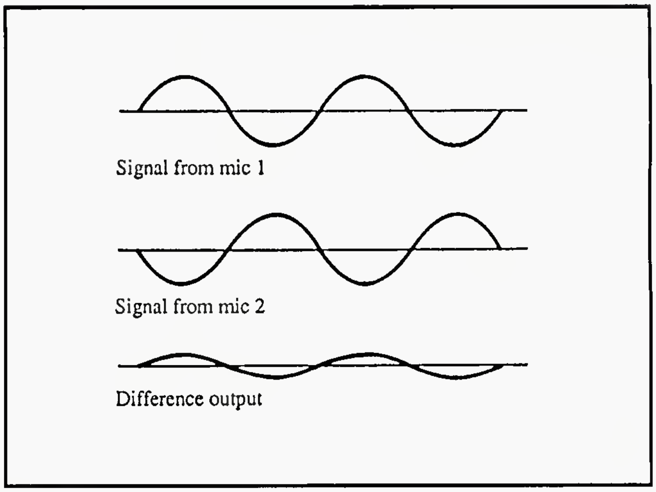


Figure 4.

Stereo television, however, does not let us off so easily. If the camera shows the kettle drums located far to the left and the French horns far to the right, we expect those sounds to be heard in the corresponding speakers. No longer can we enjoy the old FM radio luxury of having them appear in accordance with somebody's arbitrary choice. Again, in a television drama, if the doorknock comes from the right-hand speaker, we don't want to see the actor walking off to the left to answer it. Television stereo is truth time. The only exception to this are those sounds whose sources are not visible, such as unseen audiences and passing jets.

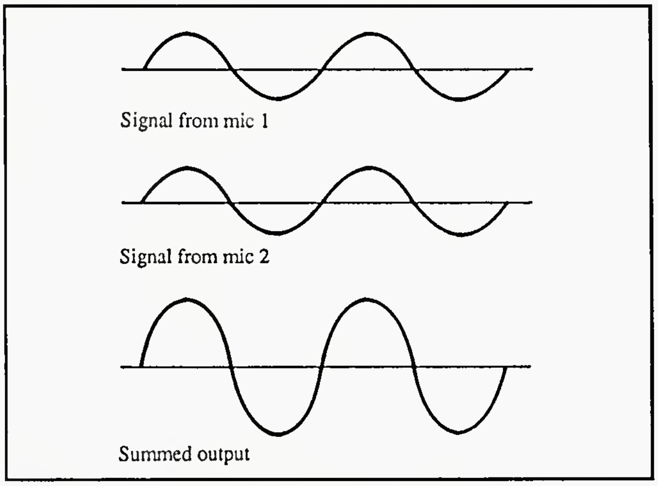


Figure 3.

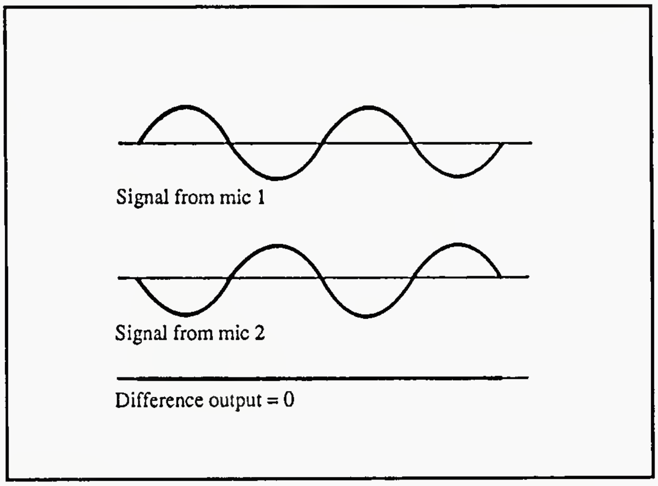


Figure 5.

As if these considerations were not enough, TV broadcasting is now offering us a treat once reserved for the theater: surround sound, a concept pioneered in large part by Dolby under the copyrighted name "Dolby Stereo." We'll look closer at this idea further on. For now, our attention is with simple stereo, left and right.

Perhaps this is not so simple after all when you consider the stiff parameters laid down by the FCC, consumer groups and the engineering fraternity itself. Foremost of these is to protect those whose sets do not receive stereo. The transmission of left and right channels cannot be allowed to degrade the normal mono signal in any way, just as "surround sound" cannot be allowed to mar the stereo feeds. Secondly, the transmitter must be capable of generating at least three other subfrequencies which will ride along with the main carrier in piggy-back fashion (a form called multiplexing) to accomplish specific jobs in specific ways. There are others, and we'll take a look at these further down the line.

Underlying most of the following discussion is the "sum and difference" principle, a concept used in a plan authorized by the FCC in 1961 for transmitting stereo in FM radio. Since 1984, it is also being used for transmitting stereo in television. In essence, the idea that if the signals sent into the right-hand channel of the audio console from the microphones on stage are both added to and subtracted from those sent into the left-hand channel, the transmitter can put out three distinct signals on only two transmission channels. Since the relatively narrow bandpass of each station in the TV spectrum allows room for only two wideband signals, a way has been found to transmit all three: mono, left audio and right audio, as shown in *Figure 1.* The term (L+R) refers, of course, to the sum of the LEFT and RIGHT microphones or channels, while (L-R) is their difference. The symbol *[x]* will be used to indicate a deliberate phase reversal.

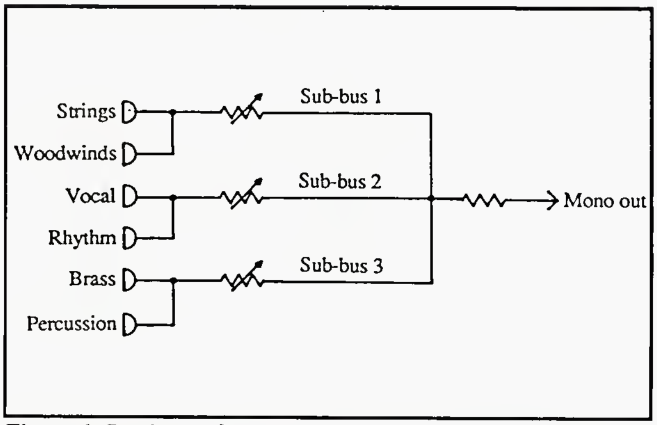


Figure 6. Sections of the orchestra are usually assigned to sub buses for the convenience of the mixer before being "summed" for a composite (mono) mix.

The circuitry for all four locations shown in *Figure 1*... the studio, the console, the transmitter and the TV receiver... has been reduced to its simplest form in this diagram to better explain its operation. In actual practice, the audio console, for example, may be capable of many more inputs and outputs than shown here, and it will no doubt have pan pots on each fader module, but

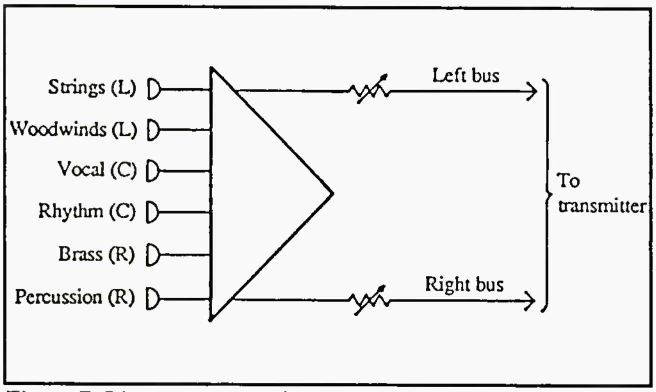
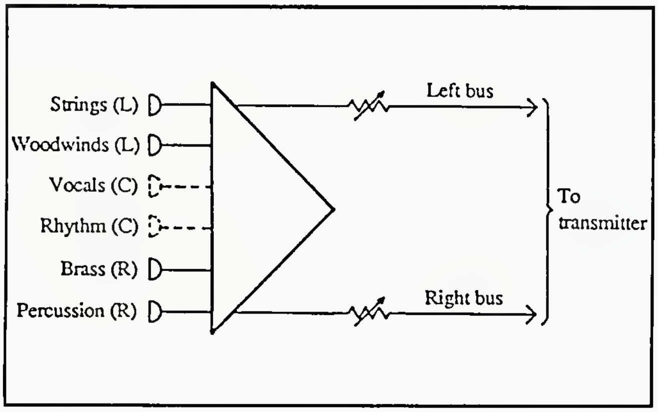


Figure 7. It's a small step from the mono mix to the stereo mix shown here. The caution lies in the monitoring, shown in Figure 9.

these conveniences are not entirely necessary for stereo operation. An assignment of the various mics and audio tape signals into either one of two channels, TOTAL LEFT or TOTAL RIGHT, is all that is needed. Many older consoles are still very much in use

Figure 8. If the center mics are fed out of phase to one stereo bus, they will not be heard in the mono output.



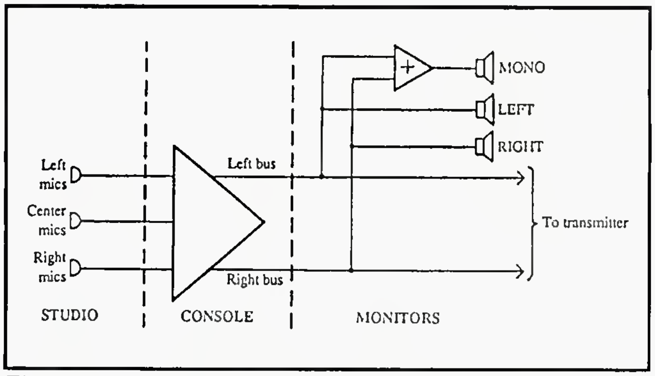
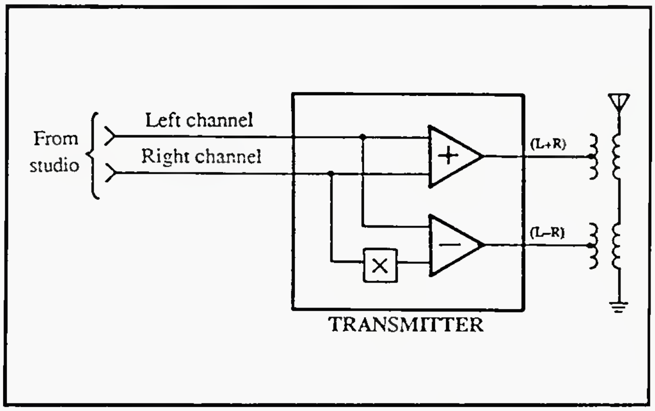


Figure 9. Even though only left and right channels are sent to the transmitter, means for monitoring the combination must be provided for the studio area, as shown here.

around the country, and the most basic illustrations here may be an advantage.

What are the sum and difference signals? The truth is that we've been dealing with them in our everyday audio work for years, perhaps without recognizing them as such. The SUM, for instance, is nothing more than the combining of any two or more signals channeled through circuits which are electrically in phase, as illustrated in *Figure 2*. Notice that it is not the external signals which must be in phase (for they seldom are, except for the briefest random moments as with this singer and his guitar), but it is the circuits carrying the signals which must be in phase. It can be said that two circuits are in phase if a common signal, fed simultaneously into each one, goes through equal values of magnitude at the same time. We make this test casually all the time when we're getting ready to use two boom mics; the mics are placed very close together and someone stands exactly between the two and utters a drawn-out "woof." If the two circuits are in phase, everything sounds fine as the signals are added together, giving us a "summed" output as shown in *Figure 3*. However, if they're out of phase, we are dealing with a "difference" signal rather than a sum. Here, the signal sounds thin and weak, or severely deteriorated as indicated in *Figure 4*. The only reason the result is not a complete cancellation is because reflections from the floor, walls, furniture, etc. cause the voice to arrive at the mics at various times. If no microphones were involved, but a pure sine wave were fed into the two circuits from an oscillator, and if the

Figure 10. The transmitter makes both sum and difference signals from the left and right channels sent from the studio, and sends these out in two separate portions of its assigned FM band.



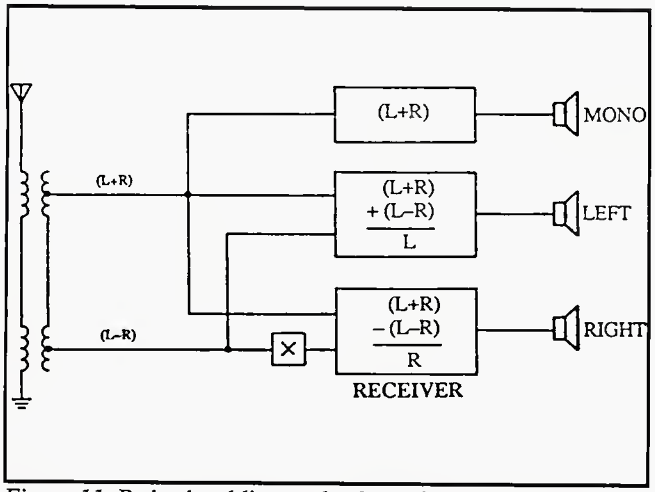


Figure 11. By both adding and subtracting the two signals coming from the transmitter, the television receiver is able to extract left and right signals from (L + R) and (L-R).

length and reactance of the circuits were identical, there would indeed be a complete cancellation of the signal as shown in *Figure 5*.

Thus, when signals of two circuits are mixed together, and one of the circuits is electrically out of phase with the other, we have "differenced" the two; in short, we have created a "difference" signal. This is normally undesirable in mono, and the solution is either to reverse the phase of one of the mics by pressing the PHASE button on the appropriate console module, to insert a phase reverser in one of the mic lines, or to reverse the #2 and #3 wires in one of the mic cables. As troublesome as this can be in normal mono operation, we'll see in a moment that deliberate phase reversal has great use in the creation of broadcast stereo.

If the singer of *Figure 2* was accompanied by a full orchestra, even in mono we normally assign the various sounds to various sub-channels for better control of the mix, as shown in *Figure 6*. We could easily change this into a stereo feed by reassigning the mics as shown in *Figure 7*.

The sounds which we want to have appear on the center channel must be assigned to both left and right channels in phase; since there is no center channel (or center speaker) in the TV receiver, we must have these sounds appear equally on the other two. This will give the illusion of a center channel to the listener. If by some chance these center mics were fed to one of the channels out of phase with the other, as indicated in *Figure 8* by the *[x]* symbol, these sounds would cancel out completely when summed at the transmitter (as described above with the tone oscillator), and will not be heard in the mono mix at all!

We take care to see that they are not fed out of phase; thus they appear properly on each channel, left and right. Even here there is a caution for the mixer: Those mics assigned to both left and right channels are being fed to the mono circuit twice! This is because the mono signal is derived from adding the left and right channels together, as shown in *Figure 1* by the summing amp in the transmitter, and shown in *Figure 9* by the summing amp in the monitor system. Even if the mix sounds fine in stereo, it may be top-heavy with center mics in mono. The difference may not be drastic, but it will be short of perfection. If the mixer prefers to monitor his work in stereo, he should at least switch his monitors over to mono once in awhile to see what he's delivering there. This may be an arguable point, but it is better to give a good mono mix and let the center channel be slightly down in stereo than to have a perfect stereo mix while the center mics are piling up in mono. As Ron Estes of NBC's The Tonight Show puts it, "You're really mixing two shows at once, and you shouldn't ignore your mono feed. So much is this a compromise that PBS has often broadcast its operas with two mixers, one doing the stereo and one mono."

The studio feeds the left and right channels to the transmitter, the detail of which is shown in *Figure 10*. Here the summing amplifier adds them together in phase by means of the familiar mixing procedure just discussed. The differential amplifier adds them out of phase by reversing the polarity of the right-hand signal as shown. The transmitter sends out the two signals simultaneously, (L + R) and (L-R).

The concept so far seems simple enough, but one might ask, "What do these two signals sound like?" The (L + R) signal is already familiar to us, for it is a mix of the two buses, left and right. In fact, it is our normal mono signal. The (L-R) signal is something else. For one thing, the center channel is missing entirely because the center mics are sent equally to both channels and reversed on one. What remains is a sound that is at best called "strange." We hear both left and right instruments, but those nearest the center (being picked up to some extent by the opposite channel) are depleted somewhat in both volume and in frequency response. This is pointed out here because it must be noted that the (L-R) signal is not a useable sound. It is merely a tool used for transmission, by which both stereo channels can be extracted in the receiver, as illustrated in *Figure 11*.

To see how this is done is only to remember our beginning algebra, and how to add and subtract positive and negative values. If (L-R) is added to (L + R) we get:

$$\begin{array}{r} (L+R) \\ +(L-R) \\ \hline 2L \end{array}$$

We can ignore the 2 since it is merely a magnitude, easily eliminated by turning the volume control down.

Similarly, if we are to subtract (L-R) from (L + R), we get:

$$\begin{array}{r} (L+R) \\ -(L-R) \\ \hline \end{array} = \begin{array}{r} (L+R) \\ +(-L+R) \\ \hline \end{array} = \begin{array}{r} (R+L) \\ +(R-L) \\ \hline 2R \end{array}$$

We drop the 2 so that (L + R) + (L-R) = L, and (L + R) - (L-R) = R. We have not only recovered our left and right channels separately (L and R), but before the arithmetic we have (L + R) which is our mono channel as well. ■

#103 TEKTEXT

JOHN G. (JAY) McKNIGHT

Demagnetizing a Tape Recorder

There seem to be more ways to demagnetize a tape recorder than there are tape recorders. So we asked Jay McKnight, the acknowledged authority on tape subjects, to set the record straight.

THE "TALKBACK" COLUMN OF DB MAGAZINE, PAGE 2 OF the 1987 March/April issue, published some answers to readers' questions about demagnetizing tape recorder heads. This brought a flood of comments and further questions to the Editor. So he asked me "is there any technical basis for a procedure for demagnetizing tape recorder heads?" Here is what we think we know.

WHAT GETS MAGNETIZED IN A TAPE RECORDER?

Anything that contains iron, nickel, or cobalt is probably "ferromagnetic," which means that it can become magnetized. But the field decreases rapidly with spacing, so our concern with magnetization can usually be limited to only those things which directly contact the tape—heads, guides, and the capstan shaft.

The materials used to make head cores and shields are very low remanence (that is, they would make terrible permanent magnets), but if they are subjected to a high magnetizing field they do retain enough magnetization to affect a tape recording.

The guides and capstan shaft are subject to the constant abrasive wear from the tape running over them, so they must be wear-resistant. What is non-magnetic and wear resistant? Ceramics are used; pyrex glass is used; but most often guides and capstan shafts are made of "non-magnetic stainless steel." As with the head material, "non-magnetic" only means that you would not intentionally use the material to make a permanent magnet. It does not mean that it has no remanence at all!

John G. (Jay) McKnight is Magnetic Reference Laboratory of Mountain View, CA 94043

Another source of unwanted magnetic field in a tape recorder could be current flowing in an electrical conductor. It takes a current of 300A to make a field of 2.5 kA/m (10 percent of the tape coercivity—the least that could affect a recording) at a distance of about 20mm away from the conductor. Such currents are never found in a tape recorder, so we won't further consider the field from a conductor. Yet more sources could be solenoids and transformers.

We have not measured their fields, but we do not believe that they produce large enough leakage fields to be a problem.

HOW DO HEADS AND GUIDES GET MAGNETIZED?

Certainly there is an obvious way to magnetize a head, guide, or capstan shaft—touch it with a permanent magnet, such as a magnetized screwdriver.

The capstan shaft runs down into the motor "works" which is a veritable hotbed of magnetic fields. They are alternating fields in many studio recorders; and even in dc motors, the magnetizing path through the capstan shaft would seem to be very inefficient. So I would not expect the motor to magnetize the capstan shaft.

You can also magnetize a head by passing direct current or a large unidirectional current pulse through it. How could this happen? Testing continuity with an ohmmeter is one way.

But tape recorders in normal use do not have magnetized screwdrivers poked into them, and the heads are not continuity tested. So how else do things get magnetized? That is one of the great mysteries of tape recording.

It may happen that an associated amplifier draws current through the head, either all of the time, or during power-up or power-down. We have had reproducing-head magnetization that seemed—as best we could find—to be caused by occasional current pulses from an integrated-circuit pre-amplifier on power-up or power-down (this was a home-built pre-amp, not a commercially-manufactured unit). This is difficult to analyze, because it happens only rarely. But once can be enough to magnetize the head.

If you find an answer, please tell me! Suffice it to say that I have observed "mysterious magnetization," and I'll bet that you have too.

SO WHO CARES ABOUT A LITTLE MAGNETIZATION ?

We all do! Magnetization can produce high second-harmonic distortion in your recordings. It can produce *clicks* at tape splices. It can produce low-frequency noise (rumblings and poppings), and high-frequency noise (hiss). It can partially erase the recorded signals—especially the short recorded wavelengths (that is, the high frequencies, especially at slower speeds).

HOW DO I KNOW IF I HAVE MAGNETIZATION ?

If magnetization is bad, you have already heard some or all of the effects just listed. But you really want to eliminate the magnetization <u>before</u> you ruin a recording session or an existing master tape, not afterward.

Finding out what is magnetized, how much it is magnetized, and why it is magnetized will take much longer than the actual demagnetizing. So if you are the usual busy maintenance engineer, you will probably just demagnetize without worrying about scientific analysis. But, for the curious ...

The list of magnetization effects can be broken in two categories—magnetization in recording, and magnetization in reproducing.

Magnetization in recording occurs because of unidirectional magnetization at a point where an alternating magnetic bias field is present. Such a point would be at the recording head, the erasing head, or both. It causes high second-harmonic distortion, high recorded background noise—mostly rumblings and poppings—and clicks at splices.

Magnetization in reproduction occurs because of unidirectional magnetization at a point where there is no alternating bias field, such as the reproducing head, guides, and capstan shaft. It causes increased noise—mostly high-frequency noise—and erasure of signals—mostly high-frequency signals.

The effects directly suggest the tests that you can perform.

RECORDING TESTS

Distortion test: Record a mid-frequency test signal (250Hz to 1000Hz) having low distortion (less than 0.2 percent second harmonic distortion) at normal recording level, using normal bias, and measure the second harmonic distortion of this recording. If the second harmonic distortion exceeds 0.5 percent, then there is probably a magnetization problem. (Note that a Total Harmonic Distortion meter is completely useless here—its indication is derived mostly from third harmonic distortion, background noise, and bias leakage. You will need a sharply tuned filter such as a wave analyzer or at least a 1/3-octave spectrometer for this measurement.)

Pop and thump test: Record a "blank" tape with normal bias. Increase the playback gain so you can clearly hear the noise. Listen to the playback of the noise. If you hear "popping and thumping" noises, you probably have a magnetization problem. This is not a very repeatable test because the "popping and thumping" are also very much a function of the blank-tape itself. So a "bad" tape on a "good" recorder may sound rather like a "good" tape on a "bad" recorder. Therefore this test may be difficult to use. I have said "listen" rather than "measure with a voltmeter" for two reasons: First, not all studios own a voltmeter that will read levels down to the -50dB to -80dB range that we are measuring here. Second, your ear can tell the difference between hiss, rumble and thumps, bias leakage, etc., but a voltmeter can't tell the difference unless you filter the noise first, and have the right averaging time in the rectifier.

Splice click test: Take the recording mentioned just previously and splice a 10mm length of non-magnetic (paper or plastic) leader into it. For best effect make a 90-degree (not 45-degree) cut; be sure the splicer is not magnetized. Play through this splice. If you hear distinct *clicks* when the splice passes the reproducing head, you have a magnetization problem. With a storage oscilloscope you could make quantitative measurements.

Reproducing tests are easy enough if the magnetized element is "downstream" of the reproducing head, because reproduction during <u>recording</u> would give a reference condition (tape has not passed the magnetized surface) but rewinding and reproducing again would give the measurement after passing the magnetized device.

Noise-increase test: Record a "blank tape" as above. Turn up the reproducer gain so you can hear the noise. First listen to the mid- and high-frequency hiss <u>as you make the recording.</u> Then rewind and play the tape several times. If the hiss increases with number of plays, then something is magnetized.

High-frequency erasure test: Record a high frequency (16kHz at 7.5 in/sec, at about - 10dB), and (as above) play during recording, then rewind and play several times. See if the recorded level remains constant for all of these playbacks. It is not unusual to see the level at this wavelength drop a few tenths of a decibel on repeated playings (attributed to magnetostriction, and commonly called "bending loss"), but if the level of the recording drops more than about 0.5dB with several plays, then magnetization is likely.

If, on the other hand, the reproducing head is magnetized, then even the first playback may be noisy and erased. In this case, you have to make a first recording that may already be noisy and erased. Then demagnetize the reproducing head, and make a second recording. Compare the noise level and high-frequency response of the first and second recordings. If they are the same, you didn't need to demagnetize the reproducing head. If the second recording has less noise and a higher level than the first, you did need to demagnetize the head.

Note that on a multi-channel recorder it is probable that the magnetization problems will be different on the different channels. So all tests have to be done on every channel that you care about. This may be a blessing in disguise, because you may find that some channels are not magnetized, and serve as reference for the magnetized channels.

MAGNETIZING AND DEMAGNETIZING

There are two problems when using a demagnetizer: first, is it strong enough to demagnetize? and second, when is it far enough away so that you can shut it off without re-magnetizing the head or guide?

Demagnetizing: If the demagnetizer is to demagnetize the core laminations, then the field that it produces must be large enough to cause the induction (flux density) in the core to approach saturation. When saturation approaches, the head output voltage waveform becomes distorted. But by this time the output voltage level is about 50dB greater than the maximum output from tape, and the playback-head pre-amplifier will surely be completely overloaded. Thus to perform this test you must disconnect the head and connect it directly to an oscilloscope input, and look for distortion in the waveform.

Now another measurement complication arises: the head output voltage is the derivative of the core flux. The effect is that when the core flux is sinusoidal, the output voltage is sinusoidal. But when the flux becomes a square wave, the head output voltage becomes a series of "spikes," and there's no way to tell just how near you are to core saturation. The "fix" for this is to build an integrating amplifier. When the head output voltage is fed through an integrator, the integrator output voltage has the same waveform as that of the core flux. Therefore the 'scope waveform will show a flat-topped wave when the core saturates.

For rough estimation purposes, you can look at the head output voltage directly on an oscilloscope. Turn the head demagnetizer on, and bring it to a point about 20mm away from the head. You should see a sinusoidal wave on the 'scope. Then bring it closer, eventually touching the core with the demagnetizer pole tips. As you bring it closer, you should come to a point where the waveform on the 'scope begins to look distorted. The sinusoid will turn into more-or-less spikes. When you begin to see spikes, the core is saturating, and the head will be magnetized if you switch the demagnetizer off, or demagnetized if you move the demagnetizer away from the head before switching it off.

Remagnetizing: So...you ask "how far away from the head do I have to be before I turn off the demagnetizer?" The magnetics text books get pretty vague here. They tell us that a field that produces less than about 10 percent of the saturation induction will not produce permanent magnetization. If you measure the actual head output voltage level at the onset of non-linearity, and compare it with the level from a test tape (at the same frequency), you will see that the level difference is about 60dB, which corresponds to an induction not 10 percent of saturation, but just one-hundredth of that, which is 0.1 percent of saturation. Put another way, it seems intuitive that the magnetization on ordinary recorded tapes will not permanently magnetize the heads, so the same field from a demagnetizer should not permanently magnetize the heads either.

Now this gets to be a measurement that you can easily do for yourself: Take a reproducer that is calibrated so that normal recording level of around 250nWb/m gives a reference (0dB) indication on the volume indicator of the reproducer. Turn on your head demagnetizer, and bring it closer to the heads, until the volume indicator reads around 0dB. (With our particular demagnetizer and heads this condition corresponded to a demagnetizer-to-head spacing of about 70mm. You will probably find a similar distance.) Now at this distance, nothing you do to turn off the demagnetizer or move it still farther away can possibly produce a high enough field to magnetize the heads.

Perhaps someone will say, "Ah, but when you switch the field off suddenly, a spike is produced." It is true that the head output voltage will produce a spike because the output voltage is the derivative (rate of change) of the magnetic field. But there is no spike in the magnetic field itself—it just falls to zero. You can confirm this for yourself. Connect a loudspeaker to the tape recorder output. Position the demagnetizer about 70mm from the reproducing head. Switch it on and off, and listen for loud clicks. You won't hear any, and this means there are no large magnetizing pulses that could magnetize the head.

CONCLUSION

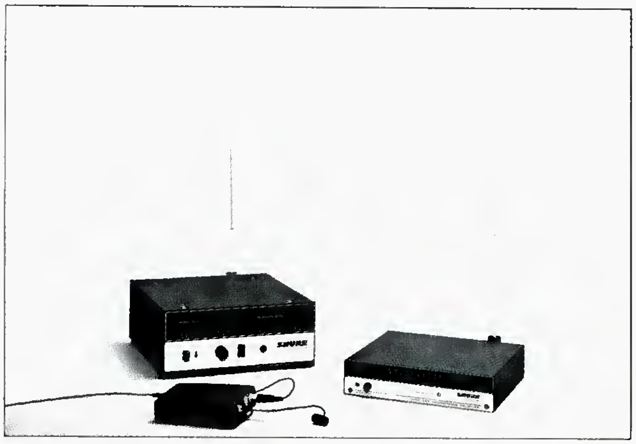
Once the demagnetizer is around 70mm away from the head, it doesn't matter how you move it around and turn it off. ■

DREW DANIELS

Hands On

The Shure Wireless Microphone Systems

Shure Brothers Inc., of Evanston, Illinois has an enviable history of innovative products that have come along as a result of staying in touch with audio equipment users and being aware of their needs. As with many audio products, they may not always be state of the art, but are so universally usable that they become *de facto* standards among users who have tried other products and found them lacking in durability or quality or usefulness. The SM58 ball microphone used by probably hundreds of thousands of performers all over the world, and the moribund Shure Vocal Master entertainer's portable P.A. system that for all its faults, finds widespread testimonial by thousands of musicians for whom it still provides service. Shure products have earned the respect of sound operators and artists far and wide.

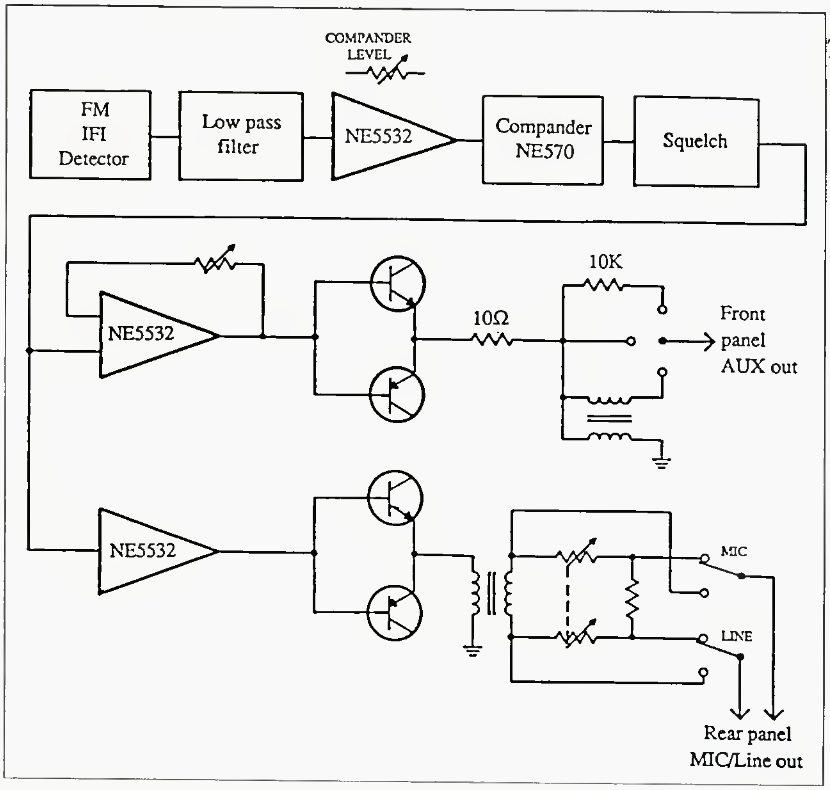


The Shure Wireless System.

The Shure Wireless system which I have examined and played with for about a month now, seems to fit the image of a device whose design and engineering were intended from the start, to endear it to the hearts of wireless users and audio engineers for years to come.

Typical of Shure products, the Shure Wireless System components are sturdily made with attention to details like the thick plating, the braking, grinding and finish of metal parts, the captive, swaged nuts on the receiver chassis, the ample mounting standoff support points for P.C. boards, the clean, mil-spec style solder connections and the use of high quality electronic parts. Thoughtful touches like additional captive nuts on the bottom of the receiver chassis and on the sides, allow the user to fix the receiver unit to any of three places between the front and rear of the unit at the bottom on both sides, or to attach rack mount panel ears without fumbling with nuts, washers and wrenches and without having to disassemble the unit.

On the inside of the W25DR receiver, the main P.C. board is clean and uncluttered, obviously designed by people who know radio design principles thoroughly. The board is a glass-epoxy type of thick stock, held down in four places by screws and captive metal standoffs and is easily removable for service or crystal changes. The highly stable soldered-in crystal should, however, quite logically only be changed by a radio technician who is completely equipped to check the retuning of the many variable coils and capacitors in the unit. The Shure factory supplies systems with any of fifteen frequencies between 167.875 and 213.8 MHz. The case of the W25DR receiver consists of the chassis, a 0.035-inch steel, brake-formed "C" channel comprising the front, bottom and back of the unit, and a second "C" channel of formed and welded 0.035-inch steel that forms the top and sides of the unit. The front panel of the unit is a 1/8-inch aluminum panel bolted by hidden swage screws through the steel chassis, and is double silk screened with dark gray and light gray paint. The dark gray portions of the panel support light gray labeling and the light gray portions support dark gray labeling showing through the lighter paint mask. The labeling should be readable in dim light, though the operation of the unit is so simple that a brief period of familiarization with the unit should enable easy operation in darkness. Four items on the front panel of the W25DR receiver, the power switch and power L.E.D., the audio output level rotary knob, an output impedance switch and a 1/4-inch phone jack, are provided for either independent headphone monitoring of the receiver's output signal, or front panel access to the output signal at a source impedance of 8 ohms, 600 ohms, or 10k ohms, for driving any load from headphones to guitar amp type high-impedance inputs. Once set, the impedance switch is stiff enough to prevent



The audio output circuits of the W25DR Receiver.

purely accidental switching if errant fingers fumble for the level control. This is good since under certain input conditions, a sudden receiver output impedance change can produce a startling and potentially dangerous volume jump in downstream equipment. Two L.E.D. bar meters at the top of the front panel indicate audio input level and RF signal level. The audio level bar meter indicates only when audio signal is present; the RF signal level bar meter indicates the RF signal strength, and is designed as an operator's guide to facilitate sound checks and rehearsals, and to indicate when transmitter-to-receiver distance is getting too great or when transmitter batteries may be draining too low. The back panel of the W25DR receiver has two UHF-type antenna connectors which accept the two supplied 5/8-wave whip antennas supplied with the unit, a screw type fuse holder, a small coaxial 13.5 volt D.C. connector, and a three-pin XL male output connector with level control and switch. This rear panel switch, labeled "MIC" and "LINE" switches the XL output connector directly to a fixed line level output amplifier or to the variable mic level control calibrated with a 0dB to-30dB scale screened onto the panel around it. With the switch in "LINE" position, the output is 600 ohms with a maximum output level of +13.5dBu. With the switch set to the "MIC" position, the output is 200 ohms with a maximum output level of -10dBu and a quoted minimum of -60dBu (though I can't quite figure out if the "minimum" means output referred to perhaps a 0VU front panel audio level reading or whatever).

Shure thoughtfully provides everything needed to set up and operate the system. Packed in the box with the W25DR receiver is a UHF antenna connector elbow for attaching one whip antenna vertically to the receiver, a 25-foot antenna cable for the second whip, and a mounting bracket for the second whip for permanent or semi-permanent installation. Even the whip antennas themselves are intelligently designed, having base collars that fit standard microphone adapters (like those for an

Figure 2. The front panel view of the W25DR Receiver.

Figure 3. A rear-panel view of the receiver.

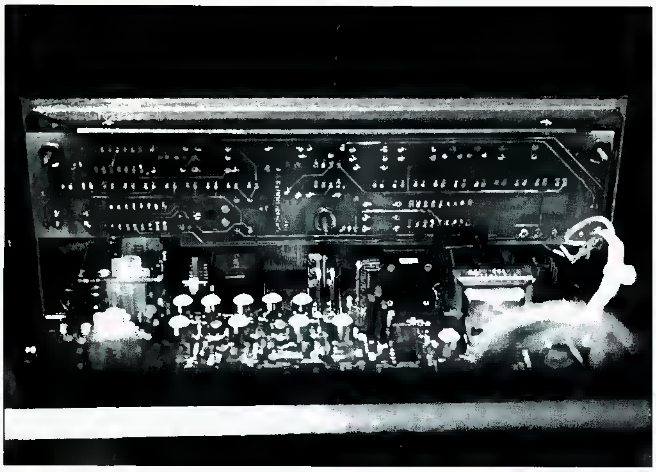


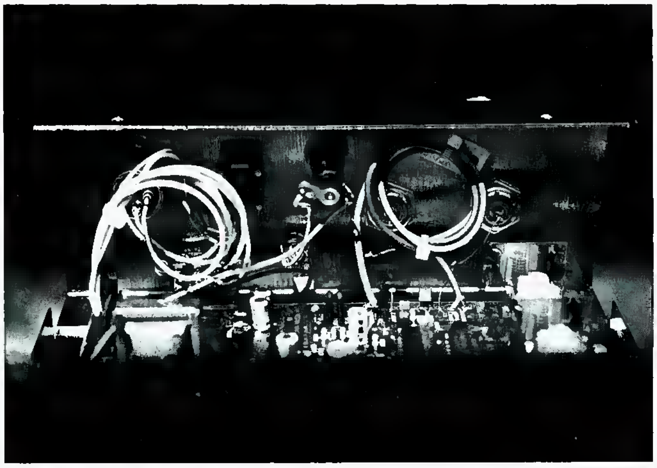
Figure 4. An inside view of the front panel..

SM58) and comprise three sections that are supplied in different lengths for different factory pre-set frequencies, and have threaded ends with heavily knurled thread extensions. The top section of the antenna rod is formed into a half-inch question mark-shaped hook which is dipped in heavy plastic insulating material so that the entire whip can be hung on a nail or from a string for maximum flexibility in setup. The permanent whip mounting

bracket is fashioned from 0.100-inch stainless steel and has a punched hole for the whip base with a slot for the cable to slip through without unscrewing the cable from the whip—great for quick daily setup and strike.

The W10BT body-pack transmitter is a sturdy, thick glass-epoxy P.C. board with neatly laid out circuits of parts soldered, again mil-spec style, into the board, standing on end, as opposed to the flat insertion of the parts in the receiver P.C. board. This on-end parts insertion style is more labor intensive, since leads must be kept short for circuit operation stability and physical ruggedness, but is absolutely necessary to enable high parts density on tiny transmitter P.C. board real estate. The case of the transmitter is made of what appears to be a cycolac-type plastic, which should provide good resistance to chipping, cracking or shattering when dropped on hard surfaces.

Figure 5. The inside view of the rear panel.



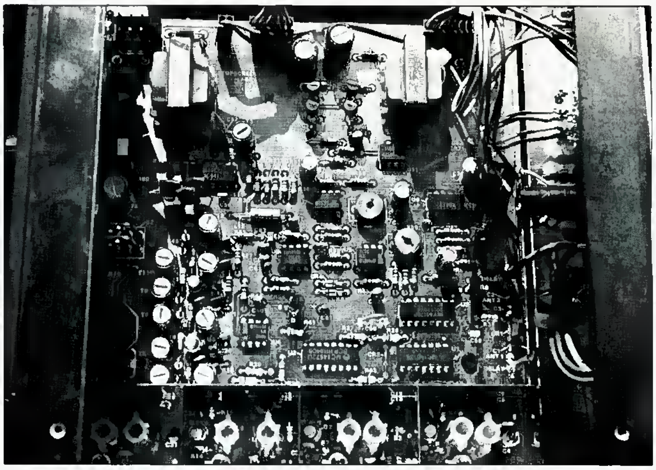
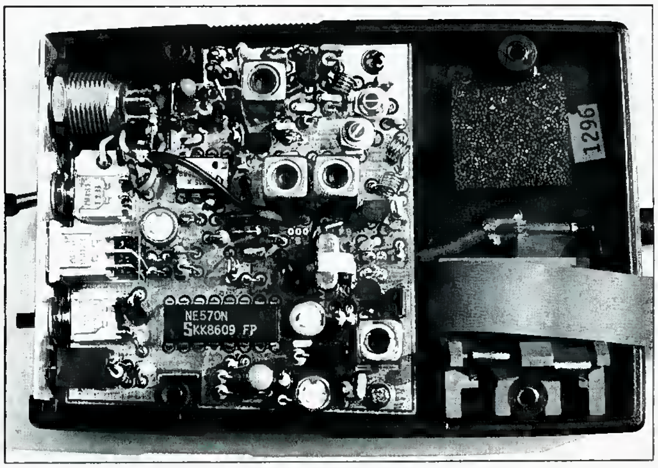


Figure 6. Inside the receiver. You can see the component board and the layout.

The W10BT transmitter is about the size of a pack of regular cigarettes or a small TV remote. It uses one 9-volt alkaline battery that will power it for 6 to 7 hours or any of several alternative batteries including Ni-Cad that will provide 1.5 to 2 hours operation on a charge. The transmitter's input connector is a Switchcraft TA4F, a four- pin miniature locking connector, located on the top of the unit as it would appear if clipped on a wearer's belt. Provided in the box with the transmitter was a cable with the mating Switchcraft connector, terminated in a 1/4-inch single circuit phone plug with a series 910k ohm resistor inside. This type of connection load is typical of a guitar amp input circuit and makes a good load for an electric guitar or bass. Other over-the-shoulder instruments like electronic keyboards will have no trouble with this type of input. The W10BT transmitter also has a power switch, a gain switch, an input signal on-off switch, a battery test switch and a battery test L.E.D. on the top of the unit. Pushing the battery check button on top of the unit lights the L.E.D. if the battery voltage is above 7.25 volts, and indicates about an hour of operation left to go with an alkaline battery. Once the unit is turned on, the associated W25DR receiver will lock onto the carrier signal and

Figure 7. The inside component board view of the transmitter.



then, as long as no other stronger radio carrier with the same frequency is detected, silence will result at that receiver's output until input signal is fed into the W10BT transmitter's input. The "MIC" switch on the W10BT simply interrupts input signal to the transmitter. The gain switch has a recessed switch handle that can be pushed with a pencil, toothpick or the small plastic screwdriver supplied with the W10BT transmitter for level adjustment. The switchable gain change is 20dB, and sets the W10BT transmitter's input so that 0.0065 volts RMS or 0.065 volts RMS will drive the modulation to 100 percent. A further, variable gain adjustment is available through a hole in the back of the W10BT transmitter's case, for which the small plastic screwdriver is actually provided. This pot has a gain control range of 20dB when the gain hi-lo switch is in the low position, and 30dB when the switch is in the high position, thus enabling a very wide range of adjustment to accommodate nearly any input likely to be plugged into the W10BT transmitter's input. The transmitter's output is 30 mW typical, 50mW maximum, and issues from a little mouse tail antenna wire that is designed to hang from the bottom of the unit. The transmitter antenna on the unit I tested was about 15-inches long and the frequency was 171MHz.

The only thing about the Shure wireless system that users might find annoying is the outboard power supply included with the system receiver. This is a 2-inch, half-pound box with an integral three-pin A.C. plug and attached cable that plugs into the receiver chassis on the rear panel. Many professional sound people have complained to various manufacturers that these outboard power supplies are inconvenient, that they must be plugged into the last socket on a power strip, or that their weight makes them fall out of the power strip in an equipment rack or that they break too easily. On the plus side, the D.C. input allows operation from automotive or battery power sources, and the low cost of the supplies, often purchased as an OEM (original equipment manufacturer) item by the equipment maker from a supplier, can, in some cases, dramatically cut design and production costs and lower the user cost of the equipment. There's also the UL listing issue. UL (Underwriter's Labs) tests electronic equipment to see if it will produce a hazard under certain adverse conditions. These conditions (e.g. overvoltage through the A.C. line to the equipment) are easier for some pieces of equipment to withstand if power supply components are removed from the main chassis.

The chassis of the receiver itself is a basic metal box with four good sized rubber feet. It is light in weight at only about 4 pounds, but there is no provision for carrying the unit. In any complete portable sound system rig, a road case with foam-lined compartments for small items like the W25DR receiver, is a necessity and in fact is standard equipment for most touring sound professionals. For me, performing occasionally in hotel ballrooms for parties or weddings, portability is important. I can normally hand carry all my equipment at once on a heavy duty folding hand truck, and the W25DR chassis slipped neatly into my equipment milk crate, the transmitter and lavalier microphone and associated connecting cables are supplied with their own Shure mic zip pouches and fit into my mic case, the whip antennas fit easily into my bass guitar gig bag with the chorus pedal, quartz tuner, strings and cords.

USING THE WIRELESS SYSTEM

I tried the system under four different conditions. The first and second tests were simply to patch the W25DR receiver into my recording console and record a voice test from the living room, then walk out the front door and down the street. A helper noted RF signal levels, tape footage and my announcements of my distance from the transmitter on my walk. Under these conditions and with only one antenna, the system provided good reception and sound quality and adequate RF signal level up to 1/4 mile from the house. With the double antenna diversity system, in a second test, the same results were achieved with no dropouts! The recorded sound of the little Shure WL83 lavalier mic was smooth and natural, not varying with distance. There were never any buzzes, hash or interference until I got out of range.

The subsequent tests were all done with a single antenna. In the second test, I played bass guitar with a casual band at the Beverly Hills Hotel. The band was off in a corner of a large dining/meeting room in the dark, and every aspect of the Shure system was effortless.

I was able to play my four sets in the dark with no buzzes, squawks or annoying sonic mishaps. This test was unusual in that my bass amp consisting of a small high-power speaker and a 400-watt MI head was set to provide about 25dB of bass and 40 dB of treble boost, thereby inviting any noises from the wireless system to be amplified to my embarrassment.

The third test was a another casual, this time singing, in Hollywood at a tennis club at the top of a hill occupied by a dozen commercial transmitters. I used a Shure SM62 dynamic mic on a short wire I made up. Performance of the Shure system was flawless. The squelch circuitry in the W25DR receiver was never challenged.

All in all, I think the Shure wireless system is the best I have seem so far. I can recommend it with no reservations. Some might find it pricey, but those who have had experience with both low cost systems and better (higher priced) systems know that the better units are simply more expensive because it cost more to produce and test high-quality radio equipment.

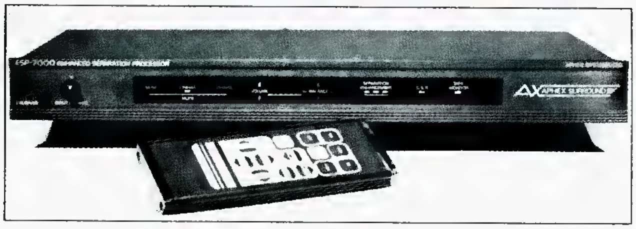
LEN FELDMAN

Lab Report

Aphex ESP-7000 Enhanced Separation Processor

GENERAL INFORMATION

Aphex Systems, Ltd. is certainly no newcomer to signal processing. Their unique products have been used in many a recording session to alter and enhance the sound quality of both vocalists and musical instruments. It is no great surprise, therefore, that the engineers at Aphex should have turned their efforts to what seems to be the most fast-moving attention getter in home and pro audio equipment—surround sound. Unlike the almost forgotten quadraphonic or four-channel implementations of the early 1970's, surround sound is justified as a home and theater entertainment medium. Many films are encoded using Dolby Stereo and that includes the surround sound idea.



The Aphex ESP-7000.

The Aphex ESP-7000, we found, delivers audiophile quality Surround Sound from virtually any stereo source, regardless of whether that signal source has been encoded or not. Sources of encoded programming include video tape cassettes, video discs, MTS or Cable TV. Non-encoded signal sources such as CDs, music videos, LP records, cassettes and FM broadcasts can also be enhanced by the signal processing circuits found in the ESP-7000.

The processor has three operating modes: Music, Cinema, and Bypass. The music mode provides up to six separate and different outputs for a full panorama of sound. Unlike reverb units, the front and surround outputs contain quite different program material. While some decoders employ time delay in addition to separation or program material between front and surround outputs, Aphex has chosen not to do so. The Music Mode is generally used when you want to create the ambience of an actual performance.

The Cinema Mode is more suited for use in conjunction with video or cinematic presentations; for the decoding of stereo audio or video programs that have been encoded with surround sound. When using the Cinema mode, frequency response and speaker assignment are done automatically to suit typical cinematic material.

Finally, the third operating mode, Stereo Bypass, eliminates all processing, as if the ESP-7000 were not in the circuit. This mode would be used for comparing the effects of the device with non-processed programming and, in some cases, for setting up initial levels, since volume and balance controls (located, by the way, on a supplied wireless remote control) can still be used in the Bypass mode.

In both the Music and Cinema modes, the ESP-7000 can provide up to 50dB of separation enhancement. More about how that is achieved in a moment. The ESP-7000 also features a subwoofer output that remains active in all three operating modes. Useful output below about 150Hz helps to improve the theater-like experience, greatly enhances music programs and aids speaker placement by allowing the use of smaller speakers for the balance of the audio spectrum.

SEPARATION ENHANCEMENT

The amount of separation enhancement available on the ESP-7000 is adjustable and, at its maximum, can provide as much as 50dB of instantaneous separation. Enhancement is also applied to the center channel output, to avoid the "pulling" effect that often occurs when a listener sits too close to either the left-front or right-front speakers in a surround sound setup.

Even with the types of enhancements already mentioned, one problem that sometimes causes difficulty in proper surround sound decoding is termed Dialogue Scatter. Highly sibilant sounds, if improperly recorded or reproduced, seem to come from all parts of a listening space rather than from the center channel, as they should. To compensate for this effect, the ESP-7000 has what they call a Dialogue Scatter Reduction circuit (DSR) that can be turned on when it is needed. The circuit, when used, is also helpful when playing noisy recordings since it is, in reality, a high-frequency "blend" circuit.

As for the separation enhancement feature mentioned earlier, it is particularly effective when decoding surround sound program material. Normally, simple decoders will have good separation between left-front and right-front

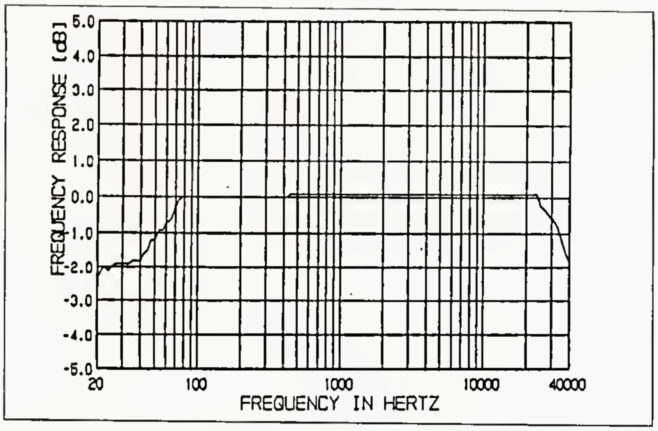


Figure 1. Front outputs frequency response plot.

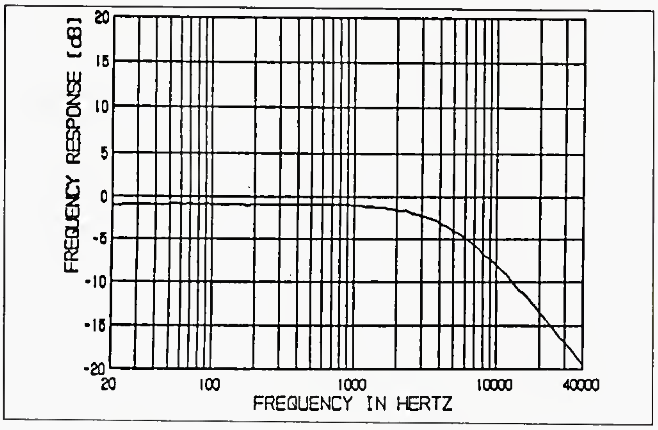


Figure 2. Rear channel plot in the Cinema mode.

channels, or between center and surround (rear) channels. However, without enhancement, separation between adjacent channels (e.g. between center channel and left front or between center channel and right front) will be only 3dB. To enhance adjacent channel separation, the Aphex ESP-7000 uses a technique known as vector cancellation, in which crosstalk components of a signal are subtracted from adjacent channels. The ESP-7000 circuits continuously measure the directional information contained in the source material and adjust the vector cancellation circuitry to yield the best instantaneous separation. Since the system is a dynamic one, it is not possible to easily measure the quantitative effect, and so we would have to take Aphex's word for the 50dB maximum separation figure. We can attest, however, to the fact that this system does work very well indeed, offering levels of apparent separation that we have not experienced with any of the several audiophile types of surround sound decoders that we have auditioned in the recent past.

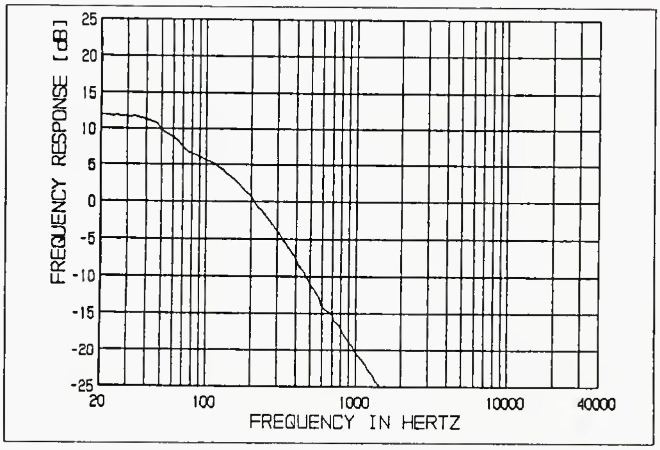
CONTROLS

Most of the control functions (including even power "on/off") for the Aphex ESP-7000 are handled by the hand-held wireless remote control. As a result, the actual slim front panel of the unit itself is largely devoted to indicator lights. The only controls found on the panel itself are a channel input balance control and a "calibrate" button. Channel balance is extremely important in any surround sound decoder and this balance control offers a range of ±6 dB. To adjust it accurately, signals containing strong center channel sound (such as dialogue) are fed to the left and right input. The "Calibrate" control is then depressed (thereby deactivating front channel outputs) and the balance control is adjusted for minimum sound heard from the surround (rear) speakers. The three modes of operation ("Bypass", "Music" and "Cinema"), status of the volume control, and status of the DSR circuit are all displayed by means of colored LEDs. Four LEDs positioned in a front-rear-left-right clock-like arrangement show setting of the relative balance of the outputs. When all four lights are of equal intensity, the balance is centered. Degree of separation enhancement is indicated by a three-LED "metering" arrangement and another indicator, near the right end of the panel, lights up when the tape monitor loop on the rear panel of the unit is being used to feed signals to the processor.

The remote control module supplied with the ESP-7000 is equipped with fourteen membrane-like momentary switches. These control power on/off of the main unit, separation enhancement amounts, mode of decoding, front/rear and left/right balance, overall volume adjustment, tape monitor loop switching and the DSR (blend) feature. In addition, an input button allows you to select one of six audio/video inputs available on an optional Aphex Remote Input Switcher if your system is equipped with that Aphex component. The remote control unit is powered by a standard 9-volt "transistor" battery.

Examining the rear panel of the unit, the two inputs are found at the extreme right of the rear panel. Stereo record-out and monitor-in jack sets comprise the tape monitor loop, while near the center of the panel there are front left and right output jacks, front-center and back-center output jacks, rear left and rear right output jacks and the sub-woofer output jack. Each of these outputs has an associated level control so that an entire system can be properly balanced for correct loudness levels from all of the speakers when you are centered in the listening room. Of course, compensation for other listening locations can then be made using the remote control's "balance" controls. There is also an accessory jack on the rear panel that connects to an Aphex Master Power Controller. If that Aphex component is installed in your system, the AC power for an entire audio/video system can be controlled when the ESP-7000 is turned on or off.

Figure 3. Response of the sub-woofer output.



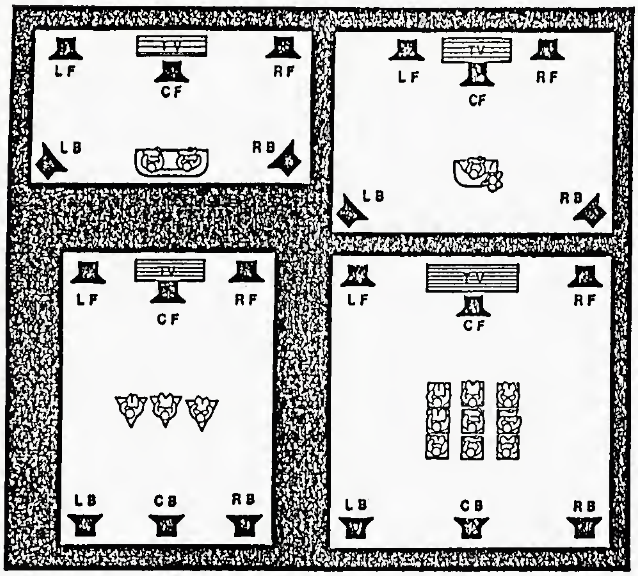


Figure 4. Suggested speaker-placement positions.

LAB MEASUREMENTS

The few relevant measurements that we were able to make for the ESP-7000 are shown in our usual table of VITAL STATISTICS at the end of this report. Distortion levels were well below significant audibility and dynamic range was 98dB, or somewhat better even than the 96dB claimed by Aphex. Frequency response of the front outputs is plotted in *Figure 1*, while response of the rear channel outputs, with the unit set to operate in the Cinema mode, is plotted in *Figure 2*. The gentle roll-off provided in this mode is generally considered desirable for rear-channel "ambience" and surround effects since, in a "live" sound experience, reverberant sounds would not have as much high frequency content as do the direct sounds. Absorbability coefficients at high frequencies are responsible for this effect, and a curve similar to that shown in *Figure 2* is not only desirable, but is typical and is commonly referred to as the "house curve." While it may vary from one concert hall to another, its general shape is not unlike that reproduced by the rear channels when in this mode.

Figure 3 illustrates the low-frequency response of the sub-woofer output. We should note that the fact that the ultra-low frequency output via this jack registered above 0dB is simply a matter of calibration. We adjusted this output for maximum simply so that we might see the full cut-off and slope characteristics of this channel. In practice, the bass level could be adjusted to whatever you require for proper balance between bass output and the rest of the program content. From all appearances, the slope of the sub-woofer output roll-off seems to be approaching a rate of 12dB per octave.

As for the gain range of the rear-panel trimmers, we measured a total end-to-end range of approximately 15dB, which is close enough to the claimed ±8dB shown in the published specifications.

COMMENTS

As noted earlier, a unit of this sort does not lend itself to many meaningful bench measurements. The true measure of its utility can only be had by hooking it into a multi-channel surround sound system and experimenting with it, exploring how effectively it decodes surround sound encoded material and how well it does with ordinary stereo programming. Despite the fact that the ESP-7000 does not sport the "Double-D" Dolby license mark, we found that the unit was compatible with Dolby encoded surround sound programming, as claimed. Mind you, the apparent sources of sound were not precisely the same as they were when using a licensed Dolby decoder, but the separation enhancement feature of the ESP-7000 more than compensated for the minor difference. If you want separation that sounds almost as good as discrete six-channel material can produce, the closest thing to that can be had using the Aphex ESP-7000 Sound Processor.

As for optimum speaker positioning, this will of course depend upon the size and shape of the listening room and its furnishing and the number of channels used. Aphex suggests several speaker arrangements, shown in *Figure 4*, one of which may prove to be perfect for your installation. As is well known, extreme bass is essntially non-directional, so if a sub-woofer is used, it can be placed almost anywhere in the listening room. Accordingly, sub-woofer placement is not included in the suggested layouts of *Figure 4*.

From a practical point of view, we would have liked to see a more precise method of indicating volume levels and balance settings than is provided by the two-LED volume indicating arrangement and the four-LED balance indicating system provided on the ESP-7000, but these are minor irritants and have nothing to do with the sound reproduction qualities of the device. We can readily understand why Aphex chose not to incorporate time-delay for the surround channels. Such time delay circuitry tends to add significant amounts of distortion to the surround output signals, unless it uses full 16-bit digital delay. And such digital delay would have made the unit far more costly than it is. As it is, the sound emanating from the rear channels was as clean and low in distortion as that coming from the front channels. There were some instances where the lack of time delay tended to de-focus the actual origin of front-channel material, but in most cases, the use of the Dialogue Scatter Reduction circuit solved that problem nicely.

In summary, what we have here is an intelligently designed surround sound processor that employs an innovative separation enhancement circuit in addition to providing an adequate number of outputs needed to recreate a true theater-like surround sound experience. We would guess that many a small recording studio or sound contractor will find any number of applications for this compact unit that takes up less than two inches of rack height. And oh, yes, incidentally, there are optional 19-inch rack mounting "ears" available so you can incorporate the ESP-7000 in your "basic" rack setup. In our auditions of the ESP-7000 we were very pleased to note that despite the really incredible apparent separation we were able to achieve with some program material, there was no obvious evidence of pumping and breathing at any time. Obviously, when it comes to signal processing, Aphex's long experience in this field has stood them in good stead. ■

VITAL STATISTICS

MAKE & MODEL: **Aphex ESP-7000 Surround Sound Processor**

SPECIFICATION	MFR'S CLAIM	db MEASURED
MAIN UNIT		
Frequency Response (Music Mode)		
Main Outputs	12 Hz-50 kHz, -1 dB	20 Hz-20 kHz,-2 dB
Sub-woofer out	12 Hz-150 Hz, -3 dB	20 Hz-150 Hz, -5 dB
Dynamic Range	96 dB	99 dB
Max. Output Level	4.2 V rms (+12.5 dBv)	Confirmed
Max. Input Level	4.2 V rms (+12.5 dBv)	Confirmed
THD for 0 dBv Input Level	0.03%	0.065%
Output Gain Trim Range	±8 dB	Confirmed
Size (HxDxW, inches)	1-3/4x9x17	Confirmed
Weight	11 lbs.	Confirmed
Power Requirements	120 VAC, 15 W	12 Watts
REMOTE CONTROL TRANSMITTER		
Size (HxWxD, inches	7/8x2-1/2x6-1/2	Confirmed
Weigh	6 ounces	Confirmed
Power Requirements	9V transistor battery	"

System Price: $995.00

Circle 60 on Reader Service Card

New Products

AURAL EXCITER

Aphex Systems Ltd. introduces its newest Aural Exciter, the Type E. The Type E is designed expressly for the performing musician. Instruments or mics can be plugged directly into the Type E for stage, recording and P.A. use without the necessity of a preamp or mixer. In fact, the Type E can serve as a low noise, high quality preamp and direct box while enhancing the sound. The Type E features "High Z" ins and outs, plus line level ins and outs, providing and even in the studio. Optional rack mounts are available for mounting one or two units in a standard 19-inch equipment rack. Like all Aural Exciters, the Type E generates musically related harmonics to restore natural clarity, detail and brightness. Aural Exciters actually recreate missing harmonics. The effect is especially helpful for digital audio effects, samplers and synthesizers because they are digitally constructing the sound which is bandwidth limited by the sampling rates.

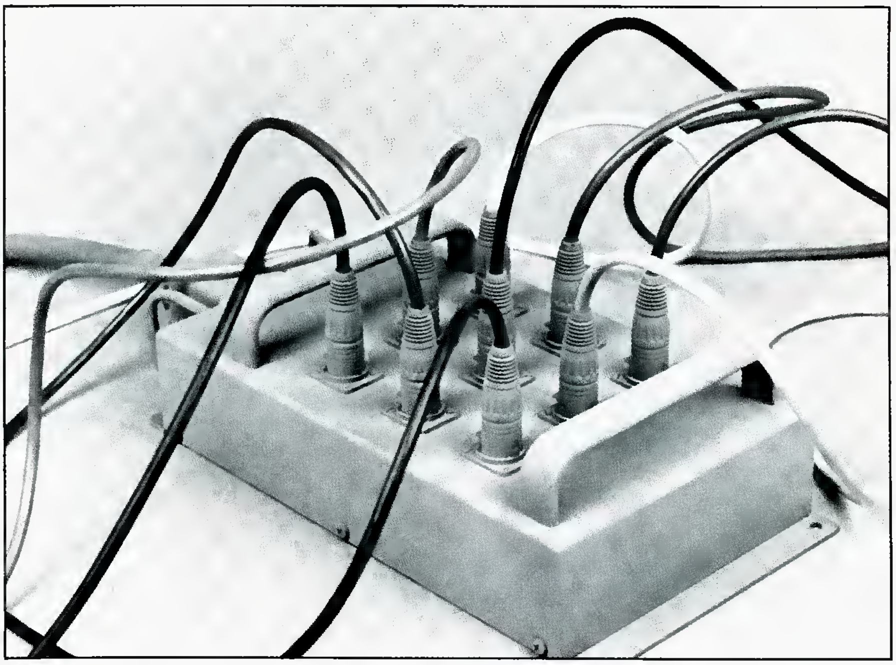
Mfr.- Aphex Systems Ltd.

Price- under $200.00

Circle 61 on Reader Service Card

MICROPHONE CABLE SERIES

Wireworks Corporation, in Hillside, New Jersey, has announced a new Microphone Cable Series which will replace the older CP Series. The new CX Series, comprising PVC jacketed, conductive plastic shielded cables, provides much more thorough shielding (100%), much greater flexibility, lower capacitance and a choice of eleven different colors. CX Series Cables are complete assemblies, standardly terminated with a Neutrik xlr compatible male and female connector on either end. They are available in six stocked lengths: 5, 10, 25, 50, 75 and 100 feet, and the following colors: black, blue, red, yellow, green, grey, pink, violet, orange, brown and white. Color is used for identification and color keying, as well as for aesthetics on stage and on camera. The particular PVC jacket featured in our CX Series is the key to the especially vibrant colors. CX Series Mic Cables, in conjunction with Wireworks' four other Mic Cable Series, are designed to complement Wireworks Professional Product Line of Audio Cabling equipment. They feature a unique lifetime warranty against defects in parts and/or workmanship.



Mfr.- Wireworks Corporation

Price: From $17.00 to $53.00

Circle 62 on Reader Service Card

DMP7 DIGITAL PROCESSOR

The Professional Audio Division of Yamaha International Corporation has introduced the DMP7 Digital Mixing Processor, an affordable digital domain mixing and processing console. The DMP7 features an 8x2 gral full-handwidth Digital Signal Processors (DSPs) in separate effects buses. Each input channel includes an A/D converter, a 3-band digital parametric equalizer, panning and effects bus control, and motorized channel faders. Effects buses 1 and 2 provide fifteen different digital effects each. Effects bus 3 offers five digital effects, an external effects send and "stereo" effects returns. The output buses feature digital compression, externally controllable output levels, D/A convertors and both balanced and unbalanced analog outputs.

All mixing and processing settings may be memorized in "snapshots" or scenes. Thirty-two snapshots may be stored on board, and another sixty-seven may be saved on the external RAM cartridge provided. A total of ninety-nine preset scenes may be instantly recalled or sequenced. Each scene may also be controlled in real time. All mixing and processing parameters may be accessed via MIDI. The DMP7's digital operating system samples at 44.1kHz, using 16-bit linear encoding. Using the digital-bus cascade feature, up to four DMP7s may be cascaded to create a 32x2 mixing format with 12 DSP multi-effects systems. Included with each DMP7 is a rack-mounting kit, an operator's manual and one external RAM cartridge. Also available as an option is the MLA7 8-channel Mic-Line Amplifier, in a single-rack-space package.

Mfr.- Yamaha

Price- $3995.00

Circle 63 on Reader Service Card

MULTI-TRACK CASSETTE/ MIXER

Fostex has introduced Model 460, a multi-track cassette/mixer capable of synchronization with video recorders or other audio recorders. The mixing section of the 460 contains 8 inputs (each with XLR-type mic connector, phantom powering, stereo send, parametric EQ and solo), 4 bus outputs, a dedicated stereo mixer for the 4-channel bus, selectable monitoring, switchable LED bargraph metering and accessible patch points for flexible system interface.

The recorder section of the 460 features a true 2-speed transport (separate record EQ circuits for 1 7/8 and 3 3/4 in./sec.), both Dolby B and C NR, 2-position autolocate, search to zero, auto repeat and SMPTE/EBU sync capability.

Mfr.- Fostex Corporation of America

Price- $2495.00

Circle 64 on Reader Service Card

People, Places... & Happenings

• **William F. Baker** has been appointed president and chief executive officer of New York metropolitan area public television station, WNET/Channel Thirteen. He succeeds **John Jay Iselin** who resigned after thirteen years as president. The announcement was made by **William M. Ellinghaus**, chairman of the board of Educational Broadcasting Corporation. Baker has been in broadcasting for twenty-five years.

• **Alpha Audio Automation Systems**, a division of **Alpha Recording Corporation**, announced an agreement with **GEXCO Technology International**, of Jersey City, New Jersey, on international marketing representation for Alpha Audio's BOSS automated audio editing system. GEXCO introduced the BOSS editor to the European market at the London Audio Engineering Society Show on March 10-13. The system shown included MIDI support, a CMX Edit Decision List Transport and Track Select on an Otari multitrack.

• **Sigma Sound Studios** has just completed a major upgrade of Philadelphia's Studio 1. A 52-input Neve 8078 recording console was installed; it is one of only three of its type ever built. It took over two years to refurbish and update the console. New features such as 6 additional effect sends and 84-input mixing capability were added while adhering to Neve specifications. The console has been fitted with a new $100,000 **George Massenburg Labs** automation system. This moving fader mixing system employs no VCAs and can store, merge and update literally thousands of mixes with frame accuracy. It can also decode mixes from Necam 96 and SSL floppies. To complete the refitting of Studio 1, Mitsubishi 2-track and 32-track digital audio recorders have also been installed, making Sigma the only 32-track digital studio in the area.

• **AKG Acoustics, Inc.**, of Stamford, Connecticut, announced that effective January 1987, it became the exclusive U.S. distributor of **Soundtracs** mixing consoles. AKG Acoustics is a wholly owned subsidiary of AKG Vienna, Austria. This announcement was made at the 81st Audio Engineering Society Convention in Los Angeles this past October. There are no immediate plans to make any major changes in the existing dealer network. AKG and Soundtracs will work together to strengthen the dealer base, increase product awareness and provide market driven products.

• Founded nearly a decade ago, **Pro Media** designs, engineers, sells, installs, rents and services professional audio, video and audio/visual products. Pro Media has been nationally acknowledged for expertise in system design, engineering installation and performance. Their staff has the ability to manage all aspects of a proposed project from the drawing board through the critical installation phase.

• **Image Express, Inc.**, the Detroit area's first full-service commercial motion-picture editing house, has hired New York editor **Steven Fineman**. After twenty-five years in New York as a commercial editor, Fineman will now be responsible for editing television commercials and documentaries in Detroit. The appointment was made by Lee Lipner, president of Image Express. The company's principal business is finishing television commercials for advertising agencies and independent producers, but it also edits documentaries, features and industrial films.

• **Valley International, Inc.** is the new name for **Valley People, Inc.** according to President **Norman Baker**. During the last eighteen months, the company has upgraded all existing products electronically and mechanically. Through research and development, nine products will be introduced which employ newly created proprietary circuitry.

• **Soundtracs Plc** has expanded both their research and development department and their manufacturing facility in Surbiton, Surrey. The research and development department, under the management of John Stadius, is responsible for designing and developing up to six new products every year. The new facility includes additional CAD design and plotting stations, ATE stations for the evaluation of new circuit designs and computer hardware for the development of digital control systems. The increased capacity of the manufacturing facility was necessary due to the demand for recently introduced models. This unit will produce the CP6800 Series and FM/FMX/FME Series of consoles, leaving more room in the other units for the PC MIDI Series.

• **Studer** has begun deliveries of its D820X two-channel DASH format digital recorder in the United States, according to **Thomas E. Mintner**, Vice President and General Manager of **Studer Revox America, Inc.** The first delivery was made on May 12 to **Disc Mastering, Inc.** in Nashville. The D820X offers an AES/EBU digital port as standard equipment, and 14-inch reel capacity allows over two hours of continuous recording time. Transport features and tape handling characteristics are equivalent to those on the A820 analog recorder. **Randy Kling**, owner and chief engineer of Disc Mastering, Inc., also took delivery of a new Neve DTC-1 digital transfer console. His facility became, for the time being, the only all-digital Studer/Neve operation in the United States.

• With deep regret and great sadness, **Audiotechniques, Inc., New York City**, announces the death of **Vice President and General Manager, Eugene Perry**. Mr. Perry, who was 38 years old, suffered a heart attack and died almost instantly. He had been with Audiotechniques for approximately four years, after being a sales manager for Harvey Radio for eight years. Mr. Perry is survived by his wife and two sons.

FROM HERE AND THERE

Over the years we have received bits of information from our readers telling us about themselves and what they do. We thought that you might be interested in knowing about them. Therefore, from time to time, we will be presenting this new column for your interest. We hope that you will find this of value, and that you, in turn, will want to tell us and all our other readers something about what you are doing in the audio field.

• West Africa is an unlikely location for a world-class recording studio. Yet it is there, in the small country of Togo, that such a studio can be found.

Equipped by 3M and valued at more than $5.2 million dollars, Togo's **Office Togolais de Disque** is described by **Gentleman's Quarterly** magazine as "Africa's best." It features a complete line of studio equipment, including a 32-input mixing console, a record pressing plant, signal processing equipment and instruments.

Designed by the British acoustics specialists **East Lake**, the studio has an impressive client list that includes international recording star **King Sunny Ade**.

In 1983, the studio produced more than 110 long-playing masters, from which 323,400 albums were printed on the premises. Record-pressing costs range from 31 to 81 cents for quantities under 500, and 15 to 56 cents for quantities of 100,000 or more.

The large, modern facility has generous office and storage space. In addition to the equipment listed above, the studio features 24- and 2-track 3M recorders, two A-700 Revox reel-to-reel tape machines, Aphex flangers and analog delay systems, electric guitars, a Hammond organ and a Rhodes piano.

Located in the capital, Lome, the studio is one of several facilities for sale or lease as part of an aggressive effort in Togo to support private enterprise and attract foreign investors.

This particular studio recently ceased operations and is being carefully maintained to keep the equipment operational. The Togolese government will offer generous financial incentives to parties interested in the purchase or lease of the facility.

Serious bidders will find the government receptive to all offers. "The government maintains a liberal investment code," the U.S. State Department reports, "providing new capital with customs duty and tax exonerations as well as the right to profit expatriation."

Known as the "Switzerland of Africa," Togo features lush beaches, state-of-the-art convention and tourist facilities and a rich blend of African/European charms.

Successful operation of the studio would require marketing expertise and possible investment in cassette duplicating equipment. Experienced recording and studio staff will be available.

• Among our international readers is **Mr. J. Lukito**. Mr. Lukito is a recording engineer for **Irama Tara Recording Studio** in Jakarta, Indonesia.

The studio produces, publishes, promotes and retails their own productions. They deal with various types of material including pop music, children's music, rock, jazz and traditional Indonesian music.

Irama Tara has a 16- and a 24-track studio. The 16-track studio utilizes a Neve console with a Studer multitrack recorder. The 24-track is comprised of a Sound Workshop Series 40 and an Otari MTR 90. Other equipment includes McIntosh and Altec power amps, JBL 4315, 4343, L100 speakers, Auratone monitor, U-47, U-87, AKG 414E 451, D-9, D-12 mics. Outboard equipment consists of Lexicon 224L, Roland 2000, 3000, Ibanez 230, Valley People Kepex, Orban De-esser, Soundcraftsmen graphic equalizer, Dolby noise reduction and Neve Compressor/Limiter. The Studer A-67 is used for mastering. The studio has the following musical instruments available: Yamaha DX-7, Solina String ensemble, Jupiter JP-8, Roland 2200 synthesizer, Hammond and Yamaha organs, Leslie speaker, Syndrum, Fender and Gibson guitars, Fender bass guitar, various guitar effects, Roger and Ludwig acoustic drums, Showman and Bassman guitar amps, Marshall speaker (Mr. Lukito added here that the guitar amp and speaker are hardly used because they usually record direct), and a Steinbach acoustic piano.

In the last two or three years, the market has been slow, so the studio rents some time out. Their rates are approximately $55.00 per shift for 16-track and $80.00 per shift for 24-track recording. The staff's shifts vary in conjunction with the schedule of the artist with whom they are currently working.

An album takes about 25 to 30 days/shifts to complete. According to Mr. Lukito, the cost of producing one's own album in Indonesia is between five and six thousand dollars. What is sold is the equivalent of an American demo tape. An artist brings the master to a cassette dealer, and if they are interested, they pay twenty-five cents per cassette jacket. The dealer makes the copy of the master and sends it to the retailer. The price of the cassette is one (U.S.) dollar. Promotion is handled through newspaper, radio and the only television station, which is owned by the government.

Mr. Lukito hoped that this gave us (and you) a brief picture of the Indonesian music business.

• **39th Street Music Productions, Inc.** in New York has recently acquired 2 Timeline Lynx Modules, in addition to the Yamaha SPX90, Yamaha DX-711FD, Yamaha 81Z and Yamaha FB-01.

• Two recording studios in South Carolina are joining forces to create a major recording center in the Southeast. **Strawberry Jamm** and **Higher Skys Studios**, both of West Columbia, South Carolina, are merging businesses to become **Strawberry Skys**, a complete 24-track, fully automated and computer assisted facility.

• **Gate Five Studios** of Sausalito, California has purchased a slew of new, top-of-the-line equipment for the studio and control room. They have recently undergone a major upgrade to implement full MIDI/SMPTE control. The upgrade features the new Soundtracs MIDI PC audio console. The studio also augmented its MIDI repertoire with several new keyboards: the Kurzweil MidiBoard controller, Roland MKS-70 Super JX module, Roland MKS-20 digital piano module, Yamaha TX-7 FM synthesizer module and Roland S-50 digital sampling synthesizer.

• **Metro Studios** of Minneapolis, Minnesota, has purchased a Drawmer DS 201x dual channel gate and a Focusrite ISA 115 HO dual mic pre-amp equalizer.

Classified

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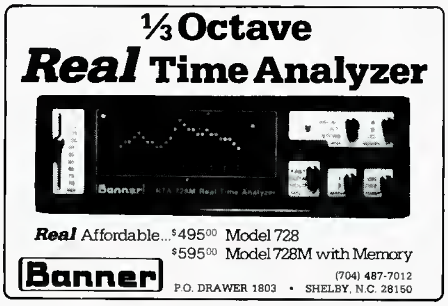

INCOMPATIBILITY? Inexpensively convert consumer equipment to professional, balanced inputs and outputs with 40 db level adjustment range and 600 ohm drive capability. For instructions, schematics and parts lists, send $4.95 to MRH AUDIO CONCEPTS, P.O. Box 339, Fox Lake, IL 60020.

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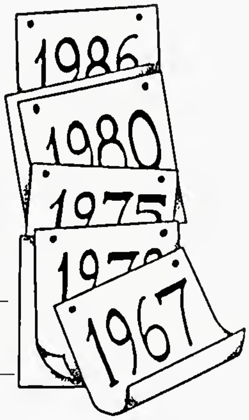

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Reverbs

ALESIS

Microverb: a 16 bit PCM professional quality reverb, 16 programs, stereo in/out, 1/4-inch jacks, -10 to +4 levels, rack mountable with Micro/Rack adapter, 6 small programs covering ambience programs, small rooms and plates, 7 large programs covering halls, chambers, plates and cavernous spaces, 1 reverse program and 2 gated reverb programs, 90dB dynamic range, 10kHz response. Input, mix and output controls. Dimensions are 5.25x6.

Price is $249.00

ART

#240 DR1 Digital Reverbation System with Performance MIDI(tm): features 21 algorithms, 40 factory plus 100 user presets. Extensive MIDI implementation via Performance MIDI(TM). 16 bit linear processor, fully software updateable. Includes remote control. Dimensions are 1.75x19x9; weight is 11 lbs.

Price $1295.00

#260 ProVerb with MIDI and Effects: has 100 presets offering: 50 Natural Reverb, 10 Gated Effects, 10 Reverse Effects, 10 Chorus Effects, 10 Echo Effects and 10 Delay Effects. It also has MIDI interface. Dimensions are 1.75x19x10; weight is 10 lbs.

Price $349.00

#230 DR2a Digital Reverb System has nine unique algorithms. It is programmable with three user presets and seven user adjustable parameters and is fully software updateable. Dimensions are 1.75x19x9; weight is 10 lbs.

Price $595.00

DIGITECH

DSP-128 has digital reverberation, chorusing, flanging, delay and delay effects,all MIDI controllable. Each program's operating parameters can be modified by the user and then stored in its original memory slot. The unit can let you have more than one effect at a time. Up to 128 different presets may be stored and accessed with MIDI program change numbers. Dimensions 1.75x19x8; weight is 4.5 lbs.

Price $399.95

FOSTEX *—See our ad on page 5*

The 3180 is a 3-spring stereo reverb with 24ms pre-delay, decay time of up to 3 seconds, RCA and 1/4-inch phone jacks, and unbalanced connectors. Dimensions are 3.5x17x8.25

Price $400.00

FURMAN SOUND

RV-3 features a choice of 2 halls, 2 rooms, 2 plates, or gated or reverse reverb. Each may be used with one of four decay times. It also features pre-delay, position, high and low rolloff buttons and direct and reverb mix controls. It has an input level control with 10-section LED meter and stereo outputs. Dimensions are 1.75x19x8 inches, weight is 8 lbs.

Price $599.00

GOTHAM *—See our ad on page 10*

EMT 246 rack mount is a digital reverberation processor and desktop control unit which features original EMT 250 reverberation program. Dimensions are 3x9.9x6.6 inches, weight is 3.3 lbs.

Price is $9,460.00

EMT 252 is a digital reverberation system using bit slice processors and multiple reverberation and effects programs. Dimensions are 3.5x8.3x12.1 inches, weight is 4.9 lbs.

Price is $15,000.00

LT SOUND *—See our ad on page 2*

RCC Reverb Control Center is a complete MicroPlate reverb system for use with or without a mixing board. It has 2 mic inputs, inputs for 2 additional stereo sources, and an output for a tape recorder. It has 3-band equalizer. Dimensions are 1.75x19x7.5; weight is 7 lbs.

Price: $595.00

RV-2 Stereo Reverb Unit features the MicroPlate reverb system and has over 18kHz of frequency response. It has 4 simultaneous inputs per channel for 3 different sounds, 7 segment LED level indicator on each channel, and decay time control of 0.6 to 2.4 seconds. Dimensions are 1.75x19x1.75; weight is 8 lbs.

Price: $895.00

ORBAN

111B reverberation is a dual-channel spring reverb with six springs/channel. The 'Floating threshold' limiter attenuates 'spring twang' and protects against overload. There is shelving bass and quasi-parametric midrange EQ. Unbalanced input accepts line-level or semi pro (medium level) gear. Transformer-balanced main output, unbalanced 'mixed' output allows use 'in-line' without external mixers. Dimensions are 3.5x19x12; weight is 10 lbs.

Price $899.00

STEVEN

PDR-3500 has 63 preset reverb sounds (50 natural reverb,9 gated effects, 4 reverse effects); MIDI interface; Input; Output mix; Level controls; Bypass switch and a wide range of algorithms. Dimensions are 1.75x19x7.5.

Price $379.00

TEAC/TASCAM *—See our ad on Cover II*

RS20B: is a dual channel spring reverb system. Features are separate in/out level controls, 200Hz HPF, 6dB per octave roll-off filter, 1/4-inch and RCA in/out jacks, effect in/out switch. Frequency response is 30Hz to 20kHz ±0.5dB reverb out, 100Hz to 4kHz +0,-6dB reverb in. S/N Ratio is 75dB (unweighted), 80dB 1Hf a WTD at 400 Hz. Dimensions are 3.6x19x9.1; weight is 9.9 lbs.

Price is $395.00

3RD GENERATION

GR2 is a stereo spring reverb with six long delay reverb springs, built-in limiters, roll off filters, variable reverb and input levels, XLR and 1/4-inch In/Out. Dimensions are 3.5x19x9, weight is 9 lbs.

Price $695.00

Delays

AKAI PROFESSIONAL

EX65D: is a 1/2 rackspace wide delay unit offering 1025ms of delay. Audio bandwidth extends to 16kHz. Modulation Width and Rate controls allow for a wide range of effects from flanging to chorusing. With the Mix and Shift outputs stereo effects can be achieved. A rear-panel "Sync" jack allows two EX65Ds to sync their modulation LFOs when used together.

Price: $329.95

ART

#250 PD-3 Professional Delay System: is a high performance, multi-tapped, digital delay system. 16 bilt A/D-D/A conversion with 64 bit kHz sampling rate. Full 20Hz to 20kHz bandwidth at all delay settings. One input, 3 outputs. One millisecond and 31.25 microsecond modes. Dimensions are 1.75x19x10; weight is 10 lbs.

Price: $749.00

AUDIO LOGIC

R2D3 Digital Room corestion Delay: Has three independent delay outputs, each capable of up to 327 milliseconds of delay. (Optional up to 1.307 seconds of delay). The unit has linear PCM 16-bit A-to-D-to-A conversion and a battery backed-up memory which retains settings when the power to the unit is removed. Minimum increment of delay is 20usecs. The unit is supplied with both XLR and barrier strip connections. Dimensions are 1.75x19x8; weight is 5 lbs.

Price: $799.00

DIGITECH

RDS 3.6 Digital Delay Unit: Offers up to 3.6 seconds of delay at half bandwidth (8kHz, 87dB) with a 10 to 1 flange ratio. The unit features chorusing, vibrato, doubling, echoes, multiple echo, flanging, feedback with flanging, comb filtering, infinite repeat. Dimensions are 1.75x19x8; weight is 7 lbs.

Price: $279.95

RDS 1900 Digital Delay Unit: Offers approximately 2 seconds of full bandwidth (15kHz, 87dB) with a 10 to 1 flange ratio. The unit features chorusing, vibrato, doubling, echoes, multiple echo, flanging, feedback with flanging, comb filtering, infinite repeat. Dimensions are 1.34x19x8; weight is 7 lbs.

Price: $289.95

RDS 2001 Digital Delay Unit: Offers up to approximately 2 seconds of full bandwidth (15kHz, 87dB) with sampling, and multiple footswitch contollable functions. The unit features 10 to 1 flange ratio, feedback with flanging, comb filtering, infinite repeat. Dimensions are 1.34x19x8; weight is 7 lbs.

Price: $309.95

RDS 7.6 Digital Delay Unit: Offers up to 7.6 seconds of delay and sample recording at full bandwidth (15kHZ, 87dB) in three ranges with a 10 to 1 flange ratio. The unit features chorusing, vibrato, doubling, echoes, multiple echo, flanging, feedback with flanging, comb filtering, infinite repeat. Dimensions are 1.75x19x8; weight is 7 lbs.

Price: $379.95

DOD ELECTRONICS CORPORATION

RD-320B Programmable Room Delay: Is a single output digital audio delay using PCM technology and is intended for large room speaker timing and concert hall spacial effects. Delay time is selected by use of dip switches through the front panel and protected by a security cover during normal operation. The delay time being equal to the sum of the time in milliseconds of the "on" switches. The time delay range is from a minimum of 5 milliseconds to a maximum in 5 millisecond increments. Dimensions are 1.75x19x6; weight is 5.75 lbs.

Price: $349.95

INDUSTRIAL RESEARCH PRODUCTS, INC.

DF-4015 Audio Signal Delay: has better than 90dB dynamic range. 192ms in 3ms steps via thumbwheel switches. 1, 2, 3, or 4 outputs. Transformer balanced in and out. Security panel. Dimensions are 1.75x19x10; weight is 12 lbs.

Price: DF-4015-1- $1598.00

DF-4015-2- $2063.00

DF-4015-3- $2528.00

DF-4015-4- $2993.00

JBL

7922 Audio Delay: has one input, two independently delayed outputs, 0 to 327 millisecond audio delay in 10 microsecond steps (about 1/8-inch sound path through air for over 350 feet). Non-volatile delay setting memory. Auto bypass maintains signal flow in case of power failures. Phase linearity; Less than 5 degrees phase rotation, 20Hz to 20kHz. Delay range; 0 to 327 milliseconds. Delay resolution; 10 microseconds/step in high resolution mode, 1 millisecond/step in low resolution mode. Digital conversion; 16 bit, linear. Sampling rate; 50kHz. Dimensions are 1.75x19x14.5; weight is 8.5 lbs.

Price: $1098.00

KLARK TEKNIK

DN-716 digital delay: 1-in, 3-out with 1.3secs. of 20Hz to 20kHz delay per output, 20μsec. increments, 90dB dynamic range, battery back-up for memories, 20dB gain trim on outputs, Ground lift, function lockout, 3 year parts warranty. Dimensions are 1.75x19x11.75; weight is 8.9 lbs.

Price: $1625.00

LT SOUND *—See our ad on page 2*

ECC Echo Control Center is a digital delay system also having Microplate reverb capability. The delay and reverb can be used together or independently. Delay times are from 1 millisecond to 1 second. Delay time on reverb is variable from 0.6 sec. to 2.4 sec.. Effects include doubling, chorus, flanging, plate reverb with delay, acoustic chamber and tremolo. Dimensions are 1.75x19x7.5.

Price: $995.00

PEAVEY ELECTRONICS CORPORATION

4530 Programmable Effects Processor: is a programmable MIDI delay unit. A four-digit, seven segment LED display provides a numeric representation of four different functional parameters of the processor. The programmable parameters are; Delay Time (milliseconds), Input Level, Feedback Level, Modulation (depth and speed), Output Mix, Sync., Fade, Filter.

Price: $699.50

SOUND CONCEPTS INC.

SSD550 Surround and Ambience Delay System: has two channels of 5ms to 50ms of delay and matrix circuit for film "surround" output. Switchable to sequential delay up to 100ms and mixed outputs available. Signal to noise ratio: 90dB response 10Hz to 8kHz. Dimensions are 3.5x19x9; weight 8 lbs.

Price: $789.00

SPECTRA SOUND

4020 Professional Audio Delay Line is designed for studio or stage use. It performs Real Time effects, has Signal Mix, Emphasis, Sweep Oscillator, and Oscillator Shape controls, from the front panel. Dimensions are 2.5x19x8.

Price: $795.00

Compressors and Limiters

ALTEC LANSING CORPORATION

1712A Compressor/Limiter: utilizes feed-forward design permitting continuously variable compression ratios from 1:1 to ∞:1. An RMS calibrated linear integration detector provides response that closely matches that of the human ear. Special compensation prevents "peak Reversion" in the detection of low frequency signals. This prevents over-compensation and audible pumping. Attack and release times are automatically adjusted. Dimensions are 1.75x19x9; weight is 6.3 lbs.

Price: $612.00

1605C NOALA (Noise Operated Automatic Level Adjustment): raises or lowers the level of a page or announcement according to the ambient noise conditions throughout the listening area at the beginning of the announcement. Dimensions are 3.5x19x7.5; weight is 12 lbs.

Price: $1916.00

APHEX SYSTEMS LTD. *—See our ad on page 11*

Compellor: delivers compression, leveling, and peak limiting simultaneously. Control circuits are analog computers that constantly monitor the input, adapt and control a single VCA per channel for minimal signal path. Features and specifications include; RF-filtered true instrumentation balanced. Output; Electronically balanced transformerless. May be operated balanced or single-ended at full output. Bandwidth; 5Hz to 65kHz $\pm$dB. Noise (referred to maximum output); -95dBm. Dynamic THD at 20dB compression, 1kHz, +4 operating level is 1% maximum. Dimensions are 1.75x19x9; weight is 9 lbs.

Price: $1195.00 (Stereo)
$795.00 (Mono)

Studio Dominator: is an intelligent 3-band peak processor with a proprietary circuit which varies the threshold for limiting. Tuneable crossover frequencies, plus high and low frequency drive controls allow the user to create different effects. Features include: Phase coherent filters in crossover, Aphex 1537A VCA control element, ALT™ Automatic Limit Threshold, TEC™ Transient Enhancement Circuit which restores the transient feel even under heavy limiting, and Servo balanced transformerless input and output stages with RF proofing. Dimensions are 1.75x19x9; weight is 8 lbs.

Price: $1195.00

ASHLY AUDIO, INC.

CL-50 Mono Limiter-Compressor

CL-52 Dual Limiter-Compressor

The CL-50, mono limiter-compressor,has a dual time constant release circuit which eliminates pumping and breathing while a soft knee threshold detector preserves subjective dynamic range. Separate adjustments for attack and release times and compression ratio allow wide-range adjustment of limiting action. The CL-52 is essentially two CL-50s in one rack space. It can be used as a stereo limiter-compressor or as two separate monos. Ten-level meters are provided on each channel to indicate gain reduction and output level.

CL-50: Dimensions are 1.75x19x6; weight is 8 lbs.

Price: $279.00

CL-52: Dimensions are 1.75x19x6; weight is 8 lbs.

Price: $479.00

AUDIO LOGIC

MT 66 Compressor/Limiter: Is a stereo compressor/limiter capable of "soft knee" dynamic range compression, or hard or soft limiting from 1:1 to ∞:1, with up to 25dB of gain reduction, and includes accessible "side chains" and a noise gate on each channel. Dimensions are 1.75x19x9; weight is 5 lbs.

Price: $299.95

BIAMP SYSTEMS

LG-2: Stereo compressor/noise gate; Threshold indicators; -40dB to +18dB threshold range; Threshold/Release controls; limiter/gate switches; Mic/Line input switching; Stereo strapping; External Trigger Inputs; Noise 76dBm; THD .01%; Frequency Response +0, -1.0dB; ground lift.

Dimensions are 1.75x19x5.5; weight is 6 lbs.

Price: $349.00

LG-4: Quad compressor/noise gate; Threshold indicators; -40dB to +18dB threshold range; Threshold/Release controls; limiter/gate switches; Mic/Line input switching; Balanced/Unbalanced XLR and 1/4-inch phone connections; Noise 76dBm; THD .01%; Frequency Response +0, -1.0dB; ground lift. Dimensions are 1.75x19x5.5; weight is 6 lbs.

Price: $499.00

BROOKE SIREN SYSTEMS

DPR-402 two channel compressor/limiter/de-esser: Wideband or frequency selective compression and de-essing; automatic or manual control of time constants; dual sidechains per channel; unique metering; frequency selective effects off rear multipin; stereo linking. Dimensions are 1.75x19x9; weight is 10 lbs.

Price: $1185.00

DPR-502 dual channel noise gate: Stereo or dual mono operation; manual or automatic control of threshold and attack time; gating or ducker; key sidechain with equalizer; 5dB-70dB range for gate-ducker. Dimensions are 1.75x19x9; weight is 10 lbs.

Price: $1200.00

CROWN INTERNATIONAL

PIP-CLP: Limiter card plugs into CROWN MICRO-TECH® LX, MACRO-TECH® stereo power amplifiers to prevent overload. Allows an additional 13dB of input signal. XLR inputs.

Price: $69.00

dbx PROFESSIONAL PRODUCTS DIVISION

160X Compressor/Limiter: gives the user a choice of OverEasy or hard-knee operation, irrespective of compression ratio. Features include; Dual rms display system: Monitors input or output with a 19-LED display. Simultaneously monitors gain reduction over a 40dB range with a 12-LED display.
Stereo-strappable/Infinity+compression provides "dynamic reversal" effects. Compression ratio continuously variable from 1:1 through infinity:1 to -1:1. Threshold variable from -40 to +20dBv; Output gain variable from -20 to +20dB; +24dBv input and output levels. Input and output connectors via a barrier strip or 1/4-inch tip-ring-sleeve phone; Provision for optional active-balanced output. Dimensions are 1.75x19x9.25.

Price: $399.00

165A Compressor/Limiter: Compression ratio continuously variable from 1:1 to infinity:1. In automatic mode, compressor attack and release times are determined by program material dynamics. In manual mode, variable attack and release rates allow the 165A to be used as a fast or slow rms-detecting limiter; PeakStop circuit prevents unwanted peaks from getting through. Separate detector input allows compression preemphasis and other effects; Each 165A is equipped with matched rms detectors for stereo-strapping operation without signal-summing errors. Analog rms meter is switchable to read input or output levels or the amount of gain reduction over a 30dB range; Active balanced input for hum and RF rejection; 24dBv input-output capability. Dimensions are 3.5x19x10.

Price: $799.00

163X Compressor/Limiter/Preamp: Three-step setup with front panel level set. Front panel Hi-Z input with rear panel gain trim. Rear panel line input and output; +18dBv maximum output level. Accessory kit for rack mounting 1 or 2 163Xes (or other dbx _63X products). Frequency response is 20Hz to 20kHz æ1dB; THD is 0.2%, maximum compression, 1kHz, 0dBv. Gain is 0-40dB, automatic, additional 0 to 20dB, adjustable, instrument input. Dimensions are 1.75x8.5x7.25.

Price: $149.00

166 Compressor/Limiter/Noise Gate: Noise gate with switchable release rate. LED shows gate operation. Variable OverEasy compressor with ∞:1 effects; PeakStop for good-sounding clipping. Side-chain monitoring for set-up of frequency dependent or anticipatory processing. Output level control, +21dBv maximum output. Threshold range: Compressor; -40 to +20dBv, Gate; +10 to -60dBv, PeakStop; 0 to +21dBv. Dimensions are 1.75x 19x8.

Price: $575.00

903 Compressor/Limiter Module: Attack rate is program-dependent; to achieve 63% gain reduction, 15ms for 10dB above threshold and 5ms for 20dB above threshold. Release rate is 120dB per second. Threshold is variable from -40 to +20dBv. Compression ratio is variable from 1:1 through à:1 to -1:1. Dimensions are 5.25x1.5x9.5.

Price: $359.00

DOD ELECTRONICS CORPORATION

R-825 Compressor/Limiter: Is a single channel compressor/limiter featuring a de-essing circuit. It can be linked to another unit for stereo operation and allows access to the signal processing side chain. Dimensions are 1.75x19x6; weight is 7 lbs.

Price: $249.95

EVENTIDE

Omnipressor®: professional quality dynamic modifier combining the characteristics of a compressor, expander, noise gate and limiter with a dynamic reversal feature. Dimensions are 3.5x19x9; weight is 8.5 lbs.

Price: $700.00

FOSTEX *–See our ad on page 5*

3070 Stereo Compressor/Limiter/Noise Gate: can be operated as two independent units or with stereo link. Each has input and output control, variable attack, release, compression ratio and gate threshold.

Price: $400.00

FURMAN SOUND

LC-X Expander/Compressor/Limiter is a unit that has three independently functional sectionsz: expander/gate, compressor/limiter/de-esser, and hard limiter. Controls include three threshold, two ratio, attack, release, and output. It features switchable LED meter, side chain jacks, bypass switch, De-Ess button, stereo interconnect, and on/off transient muting. There is an optional balanced configuration. Dimensions are 1.75x19x8 inches, weight is 7 lbs.

Price: $349.00

LC-6 Stereo Limiter/Compressor/Gate is a two channel unit that may be switched for stereo operation. Controls include input, output, compress threshold, gate threshold, attack, release, and ratio. It includes LED meters and side chain jacks, and a ground lift switch. There is an optional balanced configuration. Dimensions are 1.75x19x8 inches, weight is 7 lbs.

Price: TBA

LC-3A Limiter/Compressor includes input, output,attack, release, and ratio controls. It has an LED meter to indicate gain reduction. It includes overload and power indicators, side chain jacks, De-Ess button and ground lift switch. There is an optional balanced configuration. Dimensions are 1.75x19x8 inches, weight is 7 lbs.

Price: $249.00

GOTHAM *—See our ad on page 10*

NTP 179-170 is a compressor/expander/limiter with continuously variable controls for all parameters. It has a stereo configuration. Dimensions are 1.75x19x9.84 inches, weight is 7.7 lbs.

Price: $4,110.00

EMT 258 has noise filter with dynamic turnover frequency and expander function, compressor-like operation of the filter removes noise and increases signal intelligibility. Dimensions are 7.5x1.6x4.3 inches, weight is 2.2 lbs.

Price: $1,208.00

Neumann U 473A is a compressor/limiter/expander with stepped function controls and extremely low noise device with repeatable settings. Dimensions are 7.5x1.6x4.3 inches, weight is 2.2 lbs.

Price: $1,180.00

INDUSTRIAL RESEARCH PRODUCTS, INC.

DI-4019 Level-Matic™: has automatic level control with 10dB gain range. No audible noise or gain pumping. Compensates for loud versus soft talkers and variations in distance from the microphone. Security panel. Dimensions are 1.75x19x7.5; weight is 6 lbs.

Price: $742.00

LT SOUND *—See our ad on page 2*

CLX-2 is a feed-forward compressor/limiter incorporating the Allison EGC-101 VCA, features include simultaneous operation of both compressor and limiter.

Price: $895.00

ACC-2 is similar to the CLX-2 but has a full-featured expander as well. Also has an onboard oscillator for tremolo and stereo panning.

Price: $1250.00

SL-2 is a stereo limiter/expander with features that include simultaneous limiting and expansion functions, de-essing, stereo or independent operation.

Price: $395.00

MITSUBISHI PRO AUDIO GROUP

The Compressor-Limiter-Expander-Gate (CLEG) combines the functions of all four devices that are normally found as external components in the studio, which require patching to and from the console. The CLEG requires no external patching. Switches insert the CLEG at several different points in the mic or monitor signal path, or it can be switched to the patch bay. Variable controls adjust the amount of limiting, expansion, compression etc. Switches select the desired function, and an LED bargraph display indicates signal level for the selected mode. A further feature function of the CLEG is the LINK mode. It connects two or more CLEGs for stereo signal processing. Maximum output level: +24dBm, unity gain noise level: -90dBm, I.M.D.: .03% at unity gain, T.H.D.: .03% at unity gain. Dimensions are 5.25x1.6x8.5.

Price: $545.00

ORBAN ASSOCIATES INC

424A Gated Compressor/Limiter/De-Esser: A multi-purpose dynamic range control device with optimized, program controlled parameters. Manual adjustment of compression ratio, attack and release times, gating threshold, and de-esser sensitivity. "Output Trim" controls absolute peak level of VCA with accurate meter display. "Idle Gain" control helps prevent abrupt gain changes. Full function de-essers for sibilance control. Also available in single channel as Model 422A. Dimensions are 3.5x19x10; weight is 14 lbs.

Price: $989.00

464A Gated Leveler/Compressor/HF Limiter/Peak Clipper: A compact, two-channel unit featuring transparent leveler/compressor with variable release time and shape, high-frequency limiter with selectable pre-emphasis, and peak clipper. Push-buttons select various functions. Pop-off door conceals little-used controls: HF Limiter Pre-Emphasis, Meter Calibration, and Output Attenuator. Balanced, floating inputs and outputs. LED bargraphs display gain reduction and peak output levels simultaneously. Dimensions are 1.75x19x9.625; weight is 12 lbs.

Price: $959.00

412A Compressor/Limiter: Streamlined, cost-effective version of the 422A/424A with adjustable Attack and Release times, Compression Ratio, and Threshold. Front-panel Input and Output Attenuators. Illuminated Gain Reduction meter. Uses exclusive feedback control circuitry. User controls interact to simplify set-up. Active-balanced, floating input and output. Also available in Dual Channel/Stereo as Model 414A. Dimensions are 1.75x19x5.3; weight is 7 lbs.

Price: $425.00

PROTECH AUDIO CORP.

66303 Compressor/Limiter: Transformer Isolated in and out; 30dB adjustable gain. Switchable metering of output or compression. "Shadow" compression circuitry.

Price: $524.00

66304 Compressor/Limiter: 30dB adjustable gain. Switchable metering of output or compression.

Price: $489.00

ROCKTRON CORPORATION *– See our ad on page 8*

300 Compressor: provides three simultaneous functions; compression, peak limiting and the HUSH II single-ended noise reduction. Dimensions facilitate 19-inch standard rack mounting.

Price: $399.00

SPECTRA SONICS

601 Compressor/Limiter is a volume compressor and peak limiter modular (plug-in) card. It features independent functions while not exceeding total harmonic distortion. It is particularly effective in the control of sibilant sounds. Dimensions are 2.5x5x3/4; weight is 3 ounces.

Price: $142.00

610 "COMPLIMITER"™. This rack mounting unit performs the functions of volume-compression, and peak-limiting and has extremely low noise characteristics. It restricts extreme ranges and provides smooth dynamic action. Dimensions are 3.5x19x8.5; weight is 14 lbs.

Price: $699.00

SYMETRIX INC.

528 GATED COMPRESSOR/LIMITER: Program-controlled system analyzes incoming signals, adjusts attack and release times accordingly. Controls very wide dynamic range signals with no "pumping" or "breathing," reduces noises. Weight is 7 lbs.

Price: $495.00

522 COMPRESSOR/LIMITER/EXPANDER/GATE/DUCKER: This unit is a multi-function dynamic range processor with selectable operating mode. Ultra low distortion VCA and "soft knee" transition characteristics gives the 522 its transparency. Two channel mono or stereo operation. Weight is 9 lbs.

Price: $595.00

501 PEAK RMS COMPRESSOR/LIMITER: This two processor device has variable ratio compressor and an infinity-to-1 peak limiter. "Soft Knee" transition characteristic assures sonic integrity. Balanced and unbalanced inputs and outputs. Weight is 6 lbs.

Price: $425.00

CL-150B FAST RMS™ COMPRESSOR/LIMITER: Proprietary FAST RMS™ circuitry with "soft knee" transition characteristic yields smooth overall compression, excellent control of peaks. Selectable automatic or manual operation. Intergral de-esser controls sibilance and high frequency content. Weight is 7 lbs.

Price: $349.00

UREI

Teletronix LA-2A: is a Vacuum Tube Leveling Amplifier Compressor/Limiter. Still made in limited quality runs for fans of tube limiters. Gain reduction; up to 40dB, Distortion; 0.5% THD @ +10 dBm output, Noise; 70dB below +10dBm output level, Attack time; approx. 10usec. Dimensions are 5.25x19x8.

Price: $1346.00

LA-4A Compressor/Limiter: has a long-life LED optical attenuator. Smooth RMS detector action, selectable compression ratios, true standard volume indicator (VU), input overload indicator, simple stereo coupling. Attack time; 1 to 10 msecs. for 63% correction depending on signal waveform. Release time; 0,1 to 1 second for 63% return depending on duration of limiting. Compression ratio; 2:1, 4:1, 12:1, 20:1 switchable from front panel. Dimensions are 8.5x3.5x8; weight is 6.5 lbs.

Price: $496.00

1176LN Peak Limiter: incorporates four selectable compression ratios, attack time continuously adjustable from 20 to 800 microseconds, release time continuously adjustable from 50 milliseconds to 1.1 seconds. The unit has a high-impedance electronically balanced bridging input and balanced transformer output. Dimensions are 3.5x19x8; weight is 11 lbs.

Price: $596.00

1178 Dual Peak Limiter: is essentially two 1176LN peak limiters in a single chassis, the two limiter units are matched for stereo tracking and can be switched for operating as two independent limiters. Attack time; less than 20 usecs. for 100% recovery. Adjustable to 800usecs. with front panel control, release time; 50 msecs. minimum, 1.1 seconds maximum for 63% recovery. Adjustable from front panel control. Dimensions are 3.5x19x8; weight is 11 lbs.

Price: $896.00

VALLEY INTERNATIONAL INC.

811 Gain Brain II Variable Ratio Limiter: Distinguishes between the absolute voltage level of a signal and its loudness as perceived by the human ear. Peak Reversion Correction circuitry compensates for the discrimination against low frequency information. Dimensions are nominally 1.5x5.24; weight is 11 ounces.

Price: $420.00

816 LEVELLER limiter: Linear Integration Detection is employed to allow complex waveforms to exit the device at slightly higher absolute levels than do simple waveforms. There are no attack and release time controls on the LEVELLER as these functions are program-dependent. Automated Program Dependency circuitry dynamically optimizes the attack and release times as the program content changes. Once the desired input level is set and the output gain determined, the LEVELLER decides whether more or less "levelling" action is required and operates the threshold control. Dimensions are nominally 1.5x5.24; weight is 11 ounces.

Price: $420.00

LEVELLER two channel limiter: This two channel rack mount version of the single channel LEVELLER offers the same performance advantages of Linear Integration Detection and Automated Program Dependency circuitry. The two independent channels may be linked for processing stereo program material. The balanced, differential input section of each channel is capable of accepting -10dB, 0dB, or +4dB levels. Each of the low output sections offers variable gain to accommodate all operating levels found in recording and broadcast equipment. Dimensions are 1.75x19x8.5; weight is 7.25 lbs.

Price: $699.00

817 COMANDER Compressor/Expander: The compressor section features continuously variable threshold, attack time, ratio, and release time controls. An interactive expander is integrated with the compressor control circuitry to reduce residual noise. Linear Integration Detection and Peak Reversion Correction automatically alters release time in response to program content. Threshold/Ratio/Output Coupling computes the amount of additional gain required to maintain a constant nominal output level under varying combinations of input level, ratio, and threshold settings. Dimensions are nominally 1.5x5.24; weight is 11 ounces.

Price: $420.00

440 Limiter/Compressor/ Dynamic Sibilance Processor: Is a peak limiter, high quality compressor/expander package, and a Dynamic Sibilance Processor section, each controlling a common VCA. The compressor control section features adjustable threshold, attack time, ratio, and release time. An interactive expander control is integrated with the compressor control circuitry to reduce residual noise which would normally exist in the compression process. Dimensions are 1.75x19x8.5; weight is 7.25 lbs.

Price: $725.00

610 Dual Compressor/Expander: Offers two independent channels each consisting of a compressor and an expander section controlling a common channel VCA. The expander and compressor sections may be used interactively. In the Expanded Compression mode, the audio signal may be compressed to reduce dynamic range, and the expander may be used to reduce the residual noise which would otherwise be accentuated by the compression process. Is readily configured as an AGC device capable of delivering a constant output level for input signal levels ranging from -40dB to +24dB. Since the two independent channels can be coupled, it is ideal for sttereo applications whether AM, FM, or TV. Dimensions are 3.5x19x8.5; weight is 6.5 lbs.

Price: $995.00

YAMAHA *—See our ad on page 17*

GC2020B Stereo Compressor/Limiter: is a two channel compressor limiter noise-gate with a full 20Hz to 20kHz frequency range. Features include a link switch to permit operation as two independent channels or in linked mode to prevent loss of stereo perspective. A five segment LED display indicates amount of gain in dBs. Each channel has a variable expander gate (also called a noise gate) permitting noise during no-signal portions of the program to be eliminated. Detector in and out jacks allow the compressor to control or be controlled by external audio signals. Input and output connections are 1/4-inch phone and RCA type jacks. Dimensions are 1.75x19x8.75; weight is 6.6 lbs.

Price: $375.00

Crossovers

ALTEC LANSING

1631A is a two-way electronic crossover using plug-in modules to select crossover frequency, and configure specific equalization to provide flat power response for various horn/driver combinations. The high pass output has a level control and the low pass output has a delay adjustment of 0 to 23ms. Dimensions are 1.75x19x4.875; weight is 4.74 lbs.

Price: $522.00

ASHLY AUDIO

This series of electronic crossovers have the following details in common: Crossover points and damping are continuously adjustable and individual output stages with wide range gain adjustments will drive long cable runs and accurately match any power amplifiers. Two, three and four-way models are available in mono or stereo formats with 12 or 18dB/octave slopes. A peak overload circuit with an LED indicator monitors all critical points in the crossover and illuminates when signals are within a few decibels of clipping. The models range from the XR20/12 which is a mono 2 way, 12 dB/octave slope with dimensions that are 1.75x19x6 and is 8 lbs. to XR88/18 which is a stereo 4way, 18 db/octave slope and dimensions are 3.5x19x6 and weighs 10 lbs.

Prices from $259.00 to $699.00

AUDIO LOGIC

X-324 is a stereo three-way, stereo two-way with a mono sub woofer, or a mono four-way crossover. The unit has 18 dB/octave Butterworth filters in a state variable configuration with a switchable 40 Hz high pass filter. All connections for stereo-to-mono mode switching are internal and require no patching or rewiring. Connectors are balanced XLRs. Dimensions are 1.75x19x8, weight is 6.3 lbs.

Price: $329.95

BGW SYSTEMS

20 is an electronic crossover with stereo 2-way with mono, mono 3- or 4- way; adjustable frequencies, electronically balanced In/Out, subsonic filters, turn on time delay, toroidal power transformer, Butterworth, and 18 dB/octave filters. Dimensions are 1.75x19x11, weight is 16 lbs.

Price: $829.00

BIAMP SYSTEMS

SX-23 is a stereo 2-way, mono 3-way crossover with signal/peak indicators, high pass filters, ±12 dB gain range, 18 dB/octave, phase switching, ground lift, balanced/unbalanced XLR and 1/4-inch phone connections. Noise 85 dBm; THD 0.015%, Frequency response +0,-.1dB. Dimensions are 1.75x19x6, weight is 5 lbs.

Price: $499.00

SX-35 has the same as above except that it is a stereo 3-way, mono 5-way crossover, the noise is 80 dBm and the weight is 6 lbs.

Price: $599.00

BROOKE SIREN SYSTEMS

FDS-360 crossover is a stereo 2-way or mono 3- or 4- way with user selectable fixed frequency cards with 2nd, 3rd, or 4th order filter, integral mid-filter limiters on all bands, band edge phase adjustment and polarity switching, automute, and barrier strip insertion points. Dimensions are 1.75x19x9, weight is 10 lbs.

Price: $1025.00

BRYSTONVERMONT LIMITED

10B Crossover: is a 2-way stereo and 3-way mono unit. There are 12 crossover points for separate low pass and high pass functions. There are separate 6, 12, or 18dB slopes or separate 12 or 24dB slopes. There is a low pass +-5dB gain control. Dimensions are 1.75x19x10, weight is 9 lbs.

Price: not available at this time.

CARVIN *—See our ad on Cover III*

XC1000 crossover is an active biampable or triampable unit. It features 18 dB/octave Butterworth filters and has accurate summing characteristics. It features balanced inputs and outputs; 10 Hz-45 kHz frequency response; .01% THD, parametric filter controls sweepable from 90Hz to 16kHz. Weight is 10 lbs.

Price: $289.00

CELESTION / C-AUDIO

DF33: Stereo 2 way, stereo 3 way switchable, Frequency range 80Hz to 14kHz in two ranges, lo/mid (2way), Frequency range 1kHz to 16khz, mid/hi (3 way), Variable damping on each crossover point offering Butterworth characteristics when center, Phase reverse on mid and hi outputs, Mono bass mix of left and right, Individual output level control, Fitted with anti-tamper cover, 18dB per octave filter slope, 10k bal.-unbal. input, THD per band 0.05%. Output noise: -80dB; Maximum output: +15dB; Maximum input: +15dB; Crosstalk -70dB.

Price: $1199.00

CROWN INTERNATIONAL

FFX-2 Stereo Electronic Crossover: has user-set fixed crossover frequencies, 18dB per octave, hi-lo or hi-mid-lo filters, screw terminal connectors, fixed installations, mono subwoofer, or portable sound reinforcement. Dimensions are 1.75x19x6.5; weight is 4.3 lbs.

Price: $249.00

PIP-XOV Active Crossover Card: plugs into Crown Micro-Tech® LX (Macro-Tech®) stereo power amplifiers, mono active crossover provides bi-amp or tri-amp operation, 18dB/octave. User-set fixed crossover frequencies. Fourteen modes. Dimensions are 1-7/8x6-3/8x3-7/8; weight is 8.5 ounces.

Price: $89.00

DOD ELECTRONIC CORPORATION

R-835 Stereo/Mono Crossover: is a stereo two-way or mono three-way crossover with 18dB per octave Butterworth state variable filters. The state variable configuration insures symmetry about the crossover point generating both high-pass and low-pass outputs simultaneously. Switching from stereo to mono mode is internal and requires no patching or rewiring. Dimensions are 1.75x19x6.5; weight is 4.5 lbs.

Price: $259.95

ELECTRO-VOICE, INC.

XEQ-3 Mono 3-Way Electronic Crossover/Equalizer: incorporates fourth-order Linkwitz-Riley frequency dividing networks for 24dB per octave slope, variable time delay allows in-phase acoustic summing, plug-in EQ modules use low EQ for infrasonic filtering and MID EQ and HIGH EQ for CD horn and driver equalization, A level display for dynamic range, A level control, polarity reverse switch and mute switch for each output, rack mountable. Dimensions are 1.73x19x7.28; weight is 6.8 lbs.

Price: $716.00

XEQ-2 Mono 2-Way Electronic Crossover/Equalizer: has 18dB per octave slope, low-frequency Thiele EQ and variable time delay, plug-in crossover frequency and high-frequency CD horn EQ, high-frequency phase reverse; rack mountable. Dimensions are 1.73x19x4.9; weight is 4.74 lbs.

Price: $458.00

EX-18 Stereo 2-Way/Mono 3-Way Electronic Crossover: has variable frequency control from 100Hz to 16kHz (with 10x switch), 18dB per octave slope, flat Butterworth filters, phase switch channel controls, balanced inputs, low impedance unbalanced outputs, +22dB V high output, rack mountable. Dimensions are 1.75x19x5; weight is 4 lbs.

Price: $342.00

FOSTEX *—See our ad on page 5*

EN3020 Stereo 2 or 3 Way, Mono 4-Way Electronic Crossover: has 2-way mode subwoofer output, switchable 12 or 18dB per octave filters, and balanced and unbalanced inputs and outputs. Dimensions are 3.5x19.8.25.

Price: $699.00

FURMAN SOUND

TX-324 is a stereo 2-way,mono 3-way crossover that features 24dB/octave rolloff slopes. Field select (a Furman exclusive) allows optimizing filters for long-throw (Butterworth) or near field (Cauer). Hard limiters on each output with adjustable threshold provide speaker protection. Includes on/off transient muting, ground lift switch, in and out level controls and limit threshold indicators. There is an optional balanced configuration. Dimensions are 1.75x19x8 inches, weight is 7 lbs.

Price: $399.00

TX-424 is a stereo 3-way, mono 4- or 5-way crossover with features similar to the TX-324. Dimensions are 3.50x19x8 inches, weight is 9 lbs.

Price: $529.00

TX-524 is a stereo 4-way crossover with features similar to the TX-324. Dimensions are 3.50x19x8 inches, weight is 9 lbs.

Price: $649.00

TX-3A is a tunable crossover that is a 12dB/octave crossover that may be used for either stereo 2-way or mono 3-way applications. It includes calibrated input and output level controls, power indicator, and ground lift switch. There is an optional balanced configuration. Dimensions are 1.75x19x8 inches, weight is 7 lbs.

Price: $299.00

JBL

3100 Series Speaker-Level Frequency Dividing Networks: The passive networks use calibrated inductors, non-inductive, low ESR non-polarized capacitors and heavy duty resistors and switches. Unit features extensive use of impedance-smoothing conjugate circuits and Constant Directivity horn power response correction. Specifications are:

Model	Xover Freq.	Prog.Power	L.F. (Impedance)	H.F.
3105	7000Hz	70W	16 ohms	8 ohms
3110A	800Hz	300W	8 ohms	16 ohms
3115A	500Hz	300W	8 ohms	16 ohms
3120A	1250Hz	300W	8 ohms	16 ohms
3160	500Hz	600W	4 ohms	16 ohms

Prices: 3105- $99.00
3110A,3120A- $177.00
3160- $249.00

LT SOUND

—See our ad on page 2

ECU-2 is a stereo electronic crossover unit capable of stereo biamping as well as stereo triamping. Crossover points are continuously variable from 70Hz to 11kHz. It has 12dB/octave Butterworth filters, summed mono output for subwoofer operation, and individual phase inversion switches on mid and high bands. Dimensions are 1.75x19x7.5; weight is 6.4 lbs.

Price: $295.00

RAMSA/PANASONIC

WSP2: designed especially for use with WSA240 sub woofer system. Contains two balanced inputs, four unbalanced outputs. Output channels A and B include 3 selectable signal processing modes for use with different main loudspeakers. Two units can be racked side by side in one rack. Dimensions are 1.75x8.25x7-7/8; weight is 6 lbs.

Price: $175.00

RANE CORPORATION

AC 23 State Variable Time Correcting Crossover: Stereo 3-way/mono 4- or 5-way capability with automatic internal switching. Utilizes 24dB per octave filters with Linkwitz-Riley performance. All outputs are in phase with flat summed amplitude response. Also included are built-in 0-2ms delay circuits for electronic phase alignment of displaced drivers. 41-detent frequency selectors, individual level controls, muting circuits on all outputs, auto balanced/unbalanced/floating inputs/outputs. Dimensions are 1.75x19x5.25; weight is 5 lbs.

Price: $499.00

AC 22 State Variable Time Correcting Crossover: Stereo 2-way/mono 3-way configuration with features same as the AC 23.

Price: $389.00

SPECTRA SONICS

505 Electronic Filter: is designed for high pass, low pass, band pass and/or crossover frequencies for selective segregation of a particular frequency range. It includes power supply regulation and 18dB per octave filters, has two inputs and four outputs. In its modular card form, you choose a single, two or custom frequencies. Dimensions are 2.5x10x.75; weight is·5 ounces.

Price: $162.00 (single)
$172.00 (two)
$182.00 (custom)

3RD GENERATION

GX03 Active Crossover features 18dB roll-off, +6dB gain control for each band, 2 or 3 band stereo, 4 crossover points between bands, In/Out front and rear. Dimensions are 3.5x19x6; weight is 4 lbs.

Price: $659.00

UREI

525 Electronic Crossover is a stereo three-way or two-way, or mono 5-way or 4-way crossover with 18dB per octave slopes. Unity summing and maximally flat response. Built-in frequency counter with 1-Hz resolution. Mute switches for each output. Internal oscillator for frequency adjustment setup. Crossover filters; (4) 3rd order Butterworth, 18dB/octave. Indicators; Four-digit frequency display plus nine LEDs to show crossover points and combination. Dimensions are 3.5x19x9.75; weight is 10 lbs.

Price: $896.00

5235 Electronic Frequency Dividing Network. Dual channel, Crossover frequencies selected by plug-in circuit board. 12 or 18dB per octave filter slopes. Switchable subsonic high-pass functions. Channel isolation; 70dB, 20Hz-20kHz. Signal-to-noise ratio; 90dB, 20kHz equivalent bandwidth. Dimensions are 1.75x19x7.7; weight is 4 lbs.

Price: $450.00

MANUFACTURER'S ADDRESSES

Equalizers and other signal processing equipment buyer's guides, which space did not allow to be in this issue, will appear in the next (September/October) issue.

AKAI Professional
IMC
PO Box 2344
Ft. Worth, TX 76113-2344

Alesis
7347 Hinds Ave.
N. Hollywood, CA 91605

Altec Lansing Corp.
10500 West Reno Ave.
PO Box 26105
Oklahoma City, OK 73126

Aphex Systems Ltd.
13340 Saticoy St.
N. Hollywood, CA 92705

ART (Applied Research and Technology)
215 Tremont St.
Rochester, NY 14608

Ashly Audio, Inc.
100 Fernwood Ave.
Rochester, NY 14621

BGW Systems, Inc.
Box 5042
Hawthorne, CA 90251-5042

Biamp Systems, Inc.
PO Box 2160
Portland, OR 97208

Brooke Siren Systems/Klark-Teknik
30 B Banfi Plaza
Farmingdale, NY 11735

Brystonvermont Limited
RFD #4 Box 2255
Montpelier, VT 05663

Carvin Corp.
1155 Industrial Ave.
Escondido, CA 92025

Celestion International/C-Audio
Kuniholm Dr. Box 521
Holliston, MA 01746

Crown International
1718 W. Mishawaka Rd.
Elkhart, IN 46517

dbx
71 Chapel St.
Newton, MA 02195

DOD
Audio Logic
Digitech
5639 S. Riley St.
Salt Lake City, UT 84102

Electro-Voice, Inc.
600 Cecil St.
Buchanan, MI 49107

Eventide, Inc.
One Alsan Way
Little Ferry, NJ 07643

Fostex
14531 Blackburn Ave.
Norwalk, CA 90650

Furman Sound
30 Rich St.
Greenbrae, CA 94904

Gotham Audio Corp.
1790 Broadway
New York, NY 10019-1412

Industrial Research Products, Inc.
321 Bond St.
Elk Grove Village, IL 60007

JBL Professional/UREI
8500 Balboa Blvd.
Northridge, CA 91329

LP Music Group/Steven
160 Belmont Ave.
Garfield, NJ 07026

LT Sound
Dept. D-2
PO Box 338
Stone Mountain, GA 30086

Mitsubishi Pro Audio Group
225 Parkside Dr.
San Fernando, CA 91340

Orban Associates, Inc.
645 Bryant St.
San Francisco, CA 94107

Protech Audio Corp.
Flowerfiled Bldg. 1
St. James, NY 11780

Ramsa/Panasonic
6550 Katella Ave.
Cypress, CA 90630

Rane Corp.
6510 216th SW
Mountlake Terr., WA 98043

Rocktron Corporation
1633 Star Batt Drive
Rochester, MI 48063

Sound Concepts, Inc.
27 Newell Rd.
Brookline, MA 02146

Spectra Sonics/Spectra Sound
3750 Airport Rd.
Ogden, UT 84405

Symetrix Inc.
4211 24th Ave. W.
Seattle, WA 98199

TEAC Corp. of America
Tascam
7733 Telegraph Rd.
Montebello, CA 90640

3rd Generation
431 Hwy. 165
Voluntown, CT 06384

Valley International, Inc.
Box 40306 2817 Erica Pl.
Nashville, TN 37204

Yamaha
P.O. Box 6600
Buena Park, CA 90622

AUDIO LEVEL
RF SIGNAL LEVEL
SHURE
W25DR
POWER
AUDIO OUTPUT
DIVERSITY WIRELESS MICROPHONE RECEIVER
SHURE